

# SIP Endpoint Configuration

## Configuration examples for SIP-endpoints connected to OpenScape Office

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SEN VA SME SD 33

Siemens Enterprise Communications GmbH & Co. KG

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# 1 SIP Endpoints in OpenScape Office

## 1.1 Scope

This document describes the configuration of SIP-Endpoints used for integration of SIP features in OpenScape Office V3. For each endpoint a list of necessary configuration steps is provided as well as hints which endpoint functions are not supported by the OpenScape Office system.

It is assumed that a device comes with default factory settings. Thus only the necessary configuration steps are described. Changes of default values are NOT recommended and thus NOT mentioned in this document.

## 1.2 SIP Features in OpenScapeOffice

The following features are supported in OpenScapeOffice

### 1. Registration (Authentication)

Before establishing or receiving calls an endpoint MUST register at the system. The system expects the configured call number for registration. For security reasons it is strongly recommended to use Authentication.

### 2. Basic call

Incoming and outgoing as well as LateSDP Basic Call establishment is supported.

### 3. Name and Number presentation

For each extension a name can be configured in the system. The number and name is presented during call establishment if no restriction is activated.

*Note: The character set is limited to 7bit ASCII character set, thus no special characters are displayed (e.g. ä,ö,ü) they are replaced by " "(space) instead.*

*Note: Most SIP endpoints offer the capability to configure a terminal name which is transported in the display part of the From: header field. The terminal name is not used in the system. The name configured in the system is used instead.*

### 4. Call Waiting / Call Offer

Call Waiting/Call offer is deactivated in default, but can be activated for SIP endpoints too. (subscriber configuration)

### 5. Call Forwarding (CFU/CFB/CFNR)

Most SIP endpoints offer the capability to configure call forwarding targets.

*Note: It has to be checked that the call management rules of the system does not interfere with such an endpoint controlled forwarding. (e.g. if CFNR is configured in the system after 15 sec, endpoint controlled CF after 20 sec will not be performed)*

### 6. Hold / Retrieve / Alternate

Hold, Retrieve and alternate are supported for SIP endpoints. The system provides MOH (Music On Hold) for the held party. Local MOH in the endpoint MUST be deactivated.

### 7. Transfer (Attended/SemiAttended/Blind)

3 types of transfer are supported:

- Attended Transfer: Before the call is transferred the transferor has an established consultation call.
- SemiAttended Transfer: The Transferor goes in consultation and transfer as soon as the consulted party rings.
- Blind Transfer: Transfer is invoked out of the original call without consultation call

### 8. Message Waiting

Most SIP endpoints offer the capability to subscribe to a "Voicemail server". An endpoint which has subscribed to the MWI service receives notifications about a message left in OpenScape Office.

### 9. Distinctive ringing

Some SIP endpoints support the distinctive ringing feature, which allows to play different ringing signals for internal and external calls as well as for the recall. To make use of these features the SIP endpoint has to be configured accordingly.

### 10. Video

Video connections are supported between SIP endpoints connected to the same system and in an OpenScape network. Video connections are not supported on ITSP connections.

### 11. Codec support

In an OpenScape environment the following codecs will be used:

For gateway calls (analog/ISDN) : G711a, G711u and G729

For calls to the OpenStage HFA phones: G711a, G711u, G729 and G722

For Fax gateway calls (analog/ISDN): T.38 as well as G711-transparent (depending on the connected fax machine)

For Fax calls to OSO application: T.38

Calls to other SIP devices device dependant

### 12. DTMF support

In an OpenScape environment the recommended DTMF transport standard is RFC2833. No other method is supported (e.g. SIP Info)

## 1.3 Local SIP Features

The following features may be supported locally in a SIP phone

- Caller list
- Consultation Call
- Conference
- DoNotDisturb (DND)

Other SIP features are NOT supported in OpenScape Office.

## 1.4 Known restriction for all SIP endpoints

### 1. Keypad procedures

In general keypad procedures to invoke features (like \*1 or #1 for call forwarding) are NOT supported for SIP Endpoints

### 2. Ring back tone

During Transfer a held SIP endpoint does not get ring back tone, MOH is played instead

### 3. Forking

SIP endpoints can be configured in groups (hunt group, ring group) and thus are able to participate in some group features offered by OpenScapeOffice. Forking as it is defined in the SIP protocol is not supported.

### 4. Multiple registrations

Multiple registrations for one number are not possible.

## 1.5 Configure a SIP Endpoint in OpenScape Office

For the configuration of SIP endpoints start the Setup Wizard “Telephones / Subscribers-> IP-Telephones” :

|      | Box | Slot | Callno | Name       | Direct inward dialing | Type       | Licence Type      | License |
|------|-----|------|--------|------------|-----------------------|------------|-------------------|---------|
| Edit |     |      | 3348   | -          | -                     | No Port    | No License        |         |
| Edit |     |      | 3349   | -          | -                     | No Port    | No License        |         |
| Edit | 1   | 1    | 3360   | OS60 3360  | -                     | SIP Client | Comfort Plus User | ✓       |
| Edit | 1   | 1    | 3361   | OP410 3361 | -                     | SIP Client | Comfort User      | ✓       |
| Edit | 1   | 1    | 3362   |            | -                     | SIP Client | Comfort User      | ✓       |
| Edit | 1   | 1    | 3363   | -          | -                     | SIP Client | Comfort User      | ✓       |
| Edit | 1   | 1    | 3364   | -          | -                     | SIP Client | Comfort User      | ✓       |

For SIP endpoints the Type MUST be set to SIP Client. Add or change the following data to your needs: Callno, Name and appropriate License



With the data entered in the overview table a SIP endpoint is able to register without authentication. For safety reasons **it is strongly recommended to activate authentication** for every SIP endpoint in the system.

Press “Edit” to enter the “Change Station” page

Activate “Authentication active” and enter Password, SIP User ID and Realm.

- The Realm MUST be present and is predefined with a default string. It may be changed if your deployment needs to have a specific string here.
- The SIP User ID be present and is a string which is used during the authentication process. This could be e.g. the call number of the SIP endpoint.

The configured data must also be entered in the SIP devices as described throughout this document.

Setup - Wizards - Telephones / Subscribers - IP Telephones - Microsoft Internet Explorer

Setup - Wizards - Telephones / Subscribers - IP Telephones

### Change Station

**Parameter**

Type: SIP Client

Device Type: S0 Extension

Clip/Lin: -

Licence Type: Comfort Plus User

Licensed: Yes

Language: German

Call signaling internal:  
(Ringer pitch for internal calls): Ring type 1

Call signaling external:  
(Ringer pitch for external calls): Ring type 1

**Security**

Authentication active: ☒

Password: .....

Confirm password: .....

SIP User ID / Username: 3360

Realm: OSO-SIP

Help Abort Back OK & Next

## 2 SIP endpoint configuration examples

### 2.1 OpenStage15/40 SIP



Wiki-Page:

[http://wiki.siemens-enterprise.com/index.php/OpenStage\\_SIP](http://wiki.siemens-enterprise.com/index.php/OpenStage_SIP)

AdminGuide:

<http://apps.g-dms.com:8081/techdoc/en/P31003O1010M1000176A9/index.htm>

UserGuide:

<http://apps.g-dms.com:8081/techdoc/en/P31003S2000U132017619/index.htm>

The following steps describe the necessary configuration for the OpenStage 15/40 SIP Endpoints. The relevant configuration parameters are identical and the WBM pages are similar for both Endpoints.

Used Endpoint Software:

**OpenStage15 SIP V2 R0.41.2 SIP 100528**

**OpenStage40 SIP V2 R0.17.0 SIP 090721**

#### 2.1.1 Basic Configuration

Default Administrator password: "123456"

- Network – IP- configuration: if no DHCP is used, enter the IP network configuration parameters as used in your network.

The screenshot shows the 'IP configuration' page in the OpenScape Office web interface. The left sidebar contains a navigation menu with categories: Administrator Pages, User Pages, Logout, Admin Login, Network, System, Features, File transfer, Local functions, Speech, Authentication, Diagnostics, and Maintenance. The 'Network' category is expanded, showing options like IP configuration, Update Service (DLS), QoS, Port configuration, and LLDP-MED operation. The 'IP configuration' page has a 'change mode' link at the top. Below it, there are checkboxes for 'LLDP-MED Enabled' and 'DHCP Enabled'. A red circle highlights the following fields: 'IP address' (192.168.138.200), 'Subnet mask' (255.255.255.0), 'Default route' (192.168.138.1), and 'Primary DNS' (192.168.138.249). Other fields include 'DNS domain', 'Secondary DNS', 'Route 1 IP address', 'Route 1 gateway', 'Route 1 mask', 'Route 2 IP address', 'Route 2 gateway', 'Route 2 mask', 'VLAN discovery' (set to Manual), and 'VLAN ID'. At the bottom are 'Submit' and 'Reset' buttons.

- Network – Port- configuration: usually no changes necessary

The screenshot shows the 'Port configuration' page in the OpenScape Office web interface. The left sidebar is the same as in the previous screenshot. The 'Port configuration' page has a red circle highlighting the following fields: 'SIP server' (5060), 'SIP registrar' (5060), 'SIP gateway' (5060), and 'SIP local' (5060). Other fields include 'Backup proxy' (5060), 'RTP base' (5010), 'Download server (default)' (21), 'LDAP server', 'LAN port speed' (Automatic), 'PC port speed' (Automatic), 'PC port mode' (disabled), and 'PC port autoMDIX' (unchecked). At the bottom are 'Submit' and 'Reset' buttons.

- Date and time: For a correct time and date display enter the OpenScope Office IP-Address for SNTP-Server

## Registration & Basic Telephony

- System - System Identity – enter the Endpoint number and name

| Phone Value     | configured in OPENSCAPE OFFICE: Station->IP Clients->SIP Clients->Edit station parameters                        |
|-----------------|------------------------------------------------------------------------------------------------------------------|
| Terminal number | Call number                                                                                                      |
| Terminal name   | Optional, Phone name can only be seen in the network traces, OpenScope Office uses the name configured in system |

- System – Registration:

The screenshot shows the 'Registration' configuration page in the OpenScape Office web interface. The left sidebar contains a navigation menu with categories like Administrator Pages, User Pages, and Logout. The main content area is titled 'Registration' and contains three sections: 'SIP Addresses', 'SIP Session', and 'SIP Survivability'. In the 'SIP Session' section, the 'Server type' dropdown menu is highlighted with a red box and is set to 'Other'. The 'SIP Survivability' section has a checkbox for 'Backup registration allowed' which is checked. The 'Submit' and 'Reset' buttons are at the bottom of the form.



Attention: Server type MUST be set to **Other**, otherwise the endpoint will NOT go into service at OpenScape Office. After changing this value the phone MUST be re-started

| Phone Value           | configured in OpenScape Office:                                                                                |
|-----------------------|----------------------------------------------------------------------------------------------------------------|
| SIP server address    | IP-Address of OpenScape Office                                                                                 |
| SIP registrar address | IP-Address of OpenScape Office                                                                                 |
| SIP gateway address   | Left blank                                                                                                     |
|                       | <b>configured in OpenScape Office: Station-&gt;IP Clients-&gt;SIP Clients-&gt; Edit workpoint client data:</b> |
| Realm                 | Realm                                                                                                          |
| User ID               | Client-SIP User ID                                                                                             |
| Password              | Password                                                                                                       |

## 2.1.2 Call Forwarding

The endpoint offers

- CFB Forward on busy
- CFNR Forward on no reply
- CFU Forward all calls

The call forwarding targets must be entered first under “Forwarding Favorites”.

The image displays two screenshots of the SIP Endpoint Configuration web interface, illustrating the steps to configure call forwarding.

**Top Screenshot: Forwarding Settings**

The interface shows the **Administrator Pages** tab selected. The **Forwarding** settings are visible, with the **Forwarding Favourites** section highlighted by a red box. The settings include:

- Forward all calls allowed:** ☐ (unchecked)
- Forward on busy allowed:** ☒ (checked), to **6160**
- Forward on no reply allowed:** ☐ (unchecked), to **6140**
- No reply delay (seconds):** **16**

The **Alerts** section shows:

- Visual alerts:** ☒ (checked)
- Audible alerts:** ☒ (checked)
- Forwarding party:** **Display last** (dropdown menu)

**Bottom Screenshot: Forwarding Favourites**

The **Forwarding Favourites** configuration page is shown, with the **Destination 1** field highlighted by a red box. The destinations are:

- Destination 1:** **6160**
- Destination 2:** **6140**
- Destination 3:** **6141**
- Destination 4:** (empty)
- Destination 5:** (empty)

The **Submit** and **Reset** buttons are visible at the bottom of the form. A link labeled **back to Forwarding** is also present.

### 2.1.3 Message Waiting

The Endpoints support both un/subscribe MWI method and a waiting message is signaled in the display or with a fixed Voicemail-Key. If MWI shall be used a subscription is strongly recommended

- Features- Services : to activate subscribing for MWI support enter the OpenScape Office IP-Address

The screenshot shows the 'Services' configuration page in the OpenScape Office web interface. The left sidebar contains a menu with categories: Administrator Pages, User Pages, and Logout. Under 'Administrator Pages', there are sections for Network, System, Features, and Security. The 'Features' section is expanded, showing options like Configuration, DSS settings, Program keys, Fixed Keys, Keyset operation, and Services. The 'Services' page is active, displaying a form with fields for 'Message waiting server address' (highlighted with a red circle and containing '192.168.138.136'), 'Conference URI', 'Group pickup URI', 'Code for callback busy', 'Code for callback no reply', 'Code for callback cancel all', and 'BLF pickup code'. There are 'Submit' and 'Reset' buttons at the bottom of the form.

- Features- Configuration Voicemail number - enter the call number that will be used to establish a call to the Voicemail in case if a message is present

The screenshot shows the 'Configuration' page in the OpenScape Office web interface. The left sidebar is similar to the previous screenshot, but the 'Configuration' option under 'Features' is highlighted. The 'Configuration' page is active, displaying a form with sections for 'General' and 'Audio'. In the 'General' section, the 'Voice mail number' field is highlighted with a red circle and contains the value '71'. Other fields include 'Emergency number', 'Allow refuse' (checkbox), 'Hot/warm phone' (dropdown menu set to 'No action'), 'Hot/warm destination', 'Allow transfer on ring' (checkbox checked), 'Initial digit timer (seconds)' (text box with '30'), 'Allow uaCSTA' (checkbox), 'Server features' (checkbox), 'Not used timeout (minutes)' (dropdown menu set to '2'), 'Transfer on hangup' (checkbox), and 'Bridging enabled' (checkbox). The 'Audio' section includes 'Group pickup tone allowed' (checkbox), 'Group pickup as ringer' (checkbox), 'Group pickup visual alert' (dropdown menu set to 'Prompt'), and 'BLF alerting' (dropdown menu set to 'Beep'). There are 'Submit' and 'Reset' buttons at the bottom of the form.

- Local functions -> Message Settings

The OpenScope Office system always reports the amount of new messages (new items) to a SIP endpoint. There is no information about “urgent messages” (new urgent items) or “old messages” (old items, old urgent items). To avoid displaying useless information the “Message settings” should be configured “hidden” as shown in the following screenshot:

## 2.1.4 Distinctive Ringing

For distinctive ringing the OpenStage 15/40 Endpoints use the “info=” string received in the Alert-Info: header field.

To configure different ringing signals the “Ringer-Setting” has to be filled with one of the following strings:

1. “alert-internal” for Internal call
2. “alert-external” for External call
3. “alert-recall” for Recall (e.g., following transfer)

| Name           | Ringer sound | Pattern melody | Pattern sequence | Duration (sec) | Audible |
|----------------|--------------|----------------|------------------|----------------|---------|
| alert-internal | Ringer6.wav  | 1              | 1                | 60             | Ring    |
| alert-external | Ringer2.wav  | 2              | 2                | 60             | Ring    |
| alert-recall   | Ringer3.wav  | 2              | 2                | 60             | Ring    |
|                | Pattern      | 2              | 2                | 60             | Ring    |
|                | Pattern      | 2              | 2                | 60             | Ring    |
|                | Pattern      | 2              | 2                | 60             | Ring    |
|                | Pattern      | 2              | 2                | 60             | Ring    |
|                | Pattern      | 2              | 2                | 60             | Ring    |
|                | Pattern      | 2              | 2                | 60             | Ring    |
|                | Pattern      | 2              | 2                | 60             | Ring    |
|                | Pattern      | 2              | 2                | 60             | Ring    |
|                | Pattern      | 2              | 2                | 60             | Ring    |
|                | Pattern      | 2              | 2                | 60             | Ring    |

## 2.1.5 Known limitations and restrictions

### System provided MOH

**MR ID:** H74428 **MR Summary:** OpenStage15S doesn't play system MOH

Local MOH in the Phone MUST be deactivated. If local MOH is activated there will be a mixture of local and system provided MOH on the phone.

The screenshot shows the Siemens OpenStage 15 web interface. At the top, it displays the Siemens logo, 'OpenStage 15', and system information: 'Phone number 181', 'Phone IP address 192.168.138.200', and 'DNS name 181'. Below this is a navigation bar with 'User Pages', 'Administrator Pages', and a 'Logout' button. The left sidebar lists various configuration options under 'Configuration', including 'Incoming calls' and 'Connected calls' (which is currently selected). The main content area is titled 'Connected calls' and contains a list of settings with checkboxes: 'Allow call transfer' (checked), 'Allow call joining' (checked), 'Allow exit conference' (unchecked), 'Allow hold reminder' (checked), 'Hold reminder delay (minutes)' (set to 6), 'Allow music on hold' (unchecked and highlighted with a red rectangle), 'Allow conferences' (checked), 'Allow secure call alert' (checked), and 'Toggle associate' (unchecked). 'Submit' and 'Reset' buttons are at the bottom.

If OpenStage15 SIP V2 R0.41.2 SIP is used activation of local MOH will lead to a blocking of the phone during Retrieve

### Feature support

OpenStage SIP provides some features which are NOT supported by OpenScape Office (e.g. CallBack, Directed Pickup, ...)

To hide the CallBack feature from the UI deselect the following items

The screenshot shows the Siemens OpenStage 15 web interface, specifically the 'Outgoing calls' configuration page. The navigation bar and sidebar are the same as in the previous screenshot. The main content area is titled 'Outgoing calls' and contains a list of settings: 'Autodial delay (seconds)' (set to 6), 'Allow callback: busy' (unchecked and highlighted with a red rectangle), 'Allow callback: no reply' (unchecked and highlighted with a red rectangle), 'Allow busy when dialling' (unchecked), and 'Allow transfer on ring' (checked). 'Submit' and 'Reset' buttons are at the bottom.

Other features (e.g. directed pickup) cannot be disabled.

## 2.2 OptiPoint 410 standard S



Wiki-Page:

[http://wiki.siemens-enterprise.com/index.php/optiPoint\\_410/420\\_S](http://wiki.siemens-enterprise.com/index.php/optiPoint_410/420_S)

AdminGuide:

<http://apps.g-dms.com:8081/techdoc/en/P31003J4270M1000176A9/index.htm>

UserGuide:

<http://apps.g-dms.com:8081/techdoc/en/P31003J4270U100017619/wwwhelp/wwhimpl/js/html/wwhelp.htm?href=wwh3master.html>

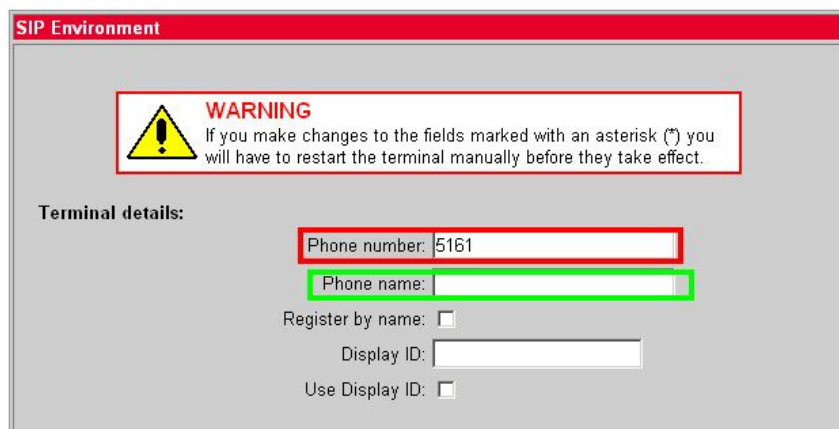
The following steps describe the necessary configuration for the OptiPoint 410 Standard S Endpoint.

Used Endpoint Software: **OptiPoint 410 S V7 R5.3.0**

### 2.2.1 Basic Configuration

Default Administrator password: "123456"

- WBM Administrator menu: **System SIP Environment**



**Terminal details:**

| Phone Value  | configured in OpenScape Office: Station->IP Clients->SIP Clients->Edit station parameters                        |
|--------------|------------------------------------------------------------------------------------------------------------------|
| Phone number | Call number                                                                                                      |
| Phone name   | Optional, Phone name can only be seen in the network traces, OpenScape Office uses the name configured in system |

All other parameters left blank

**SIP details:**

SIP routing: Server \*

Registrar IP address or DNS name: 192.168.138.135 Port: 5060

Server IP address or DNS name: 192.168.138.135 Port: 5060

Gateway IP address or DNS name: 0.0.0.0 Port: 0

SIP port: 5060

RTP Base port: 5004 \*

Outbound proxy: ☐

Default OBP domain name:

SIP transport: UDP

SIP server type: Other \*

SIP session timer enabled: ☐

SIP session timer value: 3600 seconds

Registration timer value: 3600 seconds

SIP realm: OSO135

SIP user ID: SIP-5161

New SIP password:

Confirm SIP password:

Transaction timer: 32000 milliseconds, default=32000

Registration backoff timer: 60 seconds, default=60

**Miscellaneous:**

Message Waiting IP address or DNS name: 192.168.138.135

Emergency number:

Voicemail number: 71

Branding/identity name:

**SIP details:**

| Phone Value          | configured in OpenScape Office:                                                                                |
|----------------------|----------------------------------------------------------------------------------------------------------------|
| Registrar IP address | IP-Address of OpenScape Office                                                                                 |
| Server IP address    | IP-Address of OpenScape Office                                                                                 |
| Gateway IP address   | Left blank                                                                                                     |
|                      | <b>configured in OpenScape Office: Station-&gt;IP Clients-&gt;SIP Clients-&gt; Edit workpoint client data:</b> |
| SIP realm            | Realm                                                                                                          |
| SIP user ID          | Client-SIP User ID                                                                                             |
| New SIP password     | Password                                                                                                       |

**Miscellaneous:**

| Phone Value                | configured in OpenScape Office: |
|----------------------------|---------------------------------|
| Message Waiting IP address | IP-Address of OpenScape Office  |
| Voicemail number           | Access number of VM             |

- WBM Administrator menu: **System SIP Feature:**

|                                                                               |                                                                   |
|-------------------------------------------------------------------------------|-------------------------------------------------------------------|
| <b>Call handling options</b>                                                  |                                                                   |
| Auto answer: <input type="checkbox"/> Off                                     | Auto reconnect: <input type="checkbox"/> Off                      |
| Beep on auto answer: <input checked="" type="checkbox"/> On                   | Beep on auto reconnect: <input checked="" type="checkbox"/> On    |
| Refuse call feature enabled: <input checked="" type="checkbox"/> On           | Allow transfer on ringing: <input checked="" type="checkbox"/> On |
| Allow Join in conference: <input checked="" type="checkbox"/> On              | Allow transfer on hangup: <input checked="" type="checkbox"/> On  |
| <hr/>                                                                         |                                                                   |
| <b>Group pickup</b>                                                           |                                                                   |
| Group pickup URI: <input type="text"/>                                        |                                                                   |
| <hr/>                                                                         |                                                                   |
| <b>Hot line/warm line working</b>                                             |                                                                   |
| Phone type: <input type="text" value="Deactivated"/>                          |                                                                   |
| Default dial string: <input type="text"/>                                     |                                                                   |
| <hr/>                                                                         |                                                                   |
| <b>Initial digit timer</b>                                                    |                                                                   |
| Initial digit timer value: <input type="text" value="20"/> seconds (1 to 120) |                                                                   |
| <hr/>                                                                         |                                                                   |
| <b>Station-controlled conference</b>                                          |                                                                   |
| Conference server URI: <input type="text"/>                                   |                                                                   |
| Call park URI: <input type="text" value="callpark"/>                          |                                                                   |
| Call pickup URI: <input type="text" value="pickup"/>                          |                                                                   |
| <hr/>                                                                         |                                                                   |
| <b>Callback</b>                                                               |                                                                   |
| Callback - busy URI: <input type="text"/>                                     |                                                                   |
| Callback - no reply URI: <input type="text"/>                                 |                                                                   |
| Cancel callbacks URI: <input type="text"/>                                    |                                                                   |
| <hr/>                                                                         |                                                                   |
| <b>Call Recorder</b>                                                          |                                                                   |
| Number: <input type="text"/>                                                  |                                                                   |
| Audible indication: <input type="checkbox"/> Off                              | Auto start: <input type="checkbox"/> Off                          |

These setting are left in default; the features are NOT supported by OpenScape Office

## 2.2.2 Call Forwarding

The OptiPoint 410 S Endpoints support call forwarding.

- WBM Administrator menu: **Feature Access**:  
enable “Call forwarding” Feature in the dialog box as shown in next screenshot.

**Feature access**

This page allows you to control which features are available to the User on this phone.

Checking the Status box allows the user to access the feature, but feature marked with an asterisk (\*) will also need enabling at the phone to become active.

| Feature                | Status                                |
|------------------------|---------------------------------------|
| Auto answer - CTI      | <input type="checkbox"/> *            |
| Callback - busy        | <input type="checkbox"/> *            |
| Call deflect           | <input type="checkbox"/> *            |
| Call display by number | <input checked="" type="checkbox"/> * |
| <b>Call forwarding</b> | <input checked="" type="checkbox"/> * |
| Call join              | <input type="checkbox"/> *            |
| Call pickup            | <input type="checkbox"/> *            |
| Call transfer          | <input checked="" type="checkbox"/> * |
| Do not disturb         | <input checked="" type="checkbox"/> * |
| DSM - Address book     | <input type="checkbox"/>              |
| DSM - Call control     | <input type="checkbox"/>              |
| DSM - WAP browser      | <input type="checkbox"/>              |
| Hot keypad dialing     | <input type="checkbox"/> *            |
| Log forwarded calls    | <input type="checkbox"/>              |
| Music on hold          | <input type="checkbox"/> *            |

| Feature                 | Status                                |
|-------------------------|---------------------------------------|
| Auto reconnect - CTI    | <input type="checkbox"/> *            |
| Callback - no reply     | <input type="checkbox"/> *            |
| Call display by name    | <input checked="" type="checkbox"/> * |
| Call duration           | <input checked="" type="checkbox"/> * |
| Call hold (explicit)    | <input checked="" type="checkbox"/>   |
| Call park               | <input checked="" type="checkbox"/> * |
| Call recording          | <input type="checkbox"/>              |
| Call waiting            | <input checked="" type="checkbox"/> * |
| DSM - Contacts          | <input type="checkbox"/>              |
| DSM - Speed dial        | <input type="checkbox"/>              |
| DSM - Voice recognition | <input type="checkbox"/>              |
| GPU New Call Beep       | <input type="checkbox"/>              |
| Local conference        | <input checked="" type="checkbox"/> * |
| Message waiting         | <input checked="" type="checkbox"/> * |

Note: activate and select type of forwarding at the Endpoint

### 2.2.3 Message Waiting

The OptiPoint 410 S Endpoints does NOT support the subscription to MWI service.

### 2.2.4 Distinctive Ringing

For distinctive ringing the OptiPoint 410 Endpoints use the "info=" string received in the Alert-Info: header field.

To configure different ringing signals the "Ringer-Setting" has to be filled with one of the following strings:

1. "alert-internal" for Internal call
2. "alert-external" for External call
3. "alert-recall" for Recall (e.g., following transfer)

| Alert indication | Melody<br>(0 = off,<br>or 1 to 8) | Tone<br>(1 to 3) | Duration<br>(sec) |
|------------------|-----------------------------------|------------------|-------------------|
| alert-internal   | 1                                 | 1                | 2                 |
| alert-external   | 2                                 | 2                | 5                 |
| alert-recall     | 3                                 | 3                | 10                |
|                  |                                   |                  |                   |
|                  |                                   |                  |                   |
|                  |                                   |                  |                   |
|                  |                                   |                  |                   |
|                  |                                   |                  |                   |
|                  |                                   |                  |                   |
|                  |                                   |                  |                   |
|                  |                                   |                  |                   |
|                  |                                   |                  |                   |
|                  |                                   |                  |                   |
|                  |                                   |                  |                   |
|                  |                                   |                  |                   |

Submit Reset

### 2.2.5 Known limitations and restrictions

OptiPoint 410 SIP provides some features which are NOT supported by OpenScape Office (e.g. Autoanswer, Autoreconnect, Group pickup, Hot line / warm line, Station controlled conference, CallBack, Call recorder, ...)

To hide most of these features from the UI deselect them in the [Feature Access](#) menu (see 2.2.3) and in the [System SIP Feature](#) menu (see 2.1.1)

## 2.3 Gigaset S450IP / C470IP

For more information see the Gigaset homepage:

<http://gigaset.com/hq/en/cms/PageInternetVoIPPhones.html>



The following steps describes the necessary configuration for the Gigaset S450IP/470IP End-points. The relevant configuration parameters are identical and the WBM pages are similar for both Endpoints.

Used Endpoint Software:

**Gigaset S450IP** V2 R.0.0.196 SIP 090219

**Gigaset C470IP** V2 R0.17.0 SIP 090721

### 2.3.1 Basic Configuration

Default Administrator password: "0000"

- IP Connections Parameter: if no DHCP is used, enter the IP network configuration parameters as used in your network.

Home Settings Status Log Off

**IP Configuration**

Telephony  
Messaging  
Services  
Handsets  
Miscellaneous

**Address Assignment**

IP address type: Static

IP address: 192 . 168 . 138 . 216

Subnet mask: 255 . 255 . 255 . 000

Default gateway: 192 . 168 . 138 . 249

Preferred DNS server: 192 . 168 . 138 . 249

Alternate DNS server: . . . .

**Remote Management**

Allow access from other networks: ☒ Yes ☐ No

Activating this parameter increases the risk of unauthorised access to your device settings.

**HTTP proxy**

Enable proxy: ☐ Yes ☒ No

Proxy server address:

Proxy server port: 0

Set Cancel

- Telephony->Connections (select provider)

Gigaset offers the possibility to connect to several SIP providers. OpenScape Office has to be configured as one provider. First "Edit" the data (see below) and then set the provider to "Active"

|    | Name / Provider                  | Suffix | Status     | Active                              |
|----|----------------------------------|--------|------------|-------------------------------------|
| 1. | BocaR.131-1362<br>Other Provider | #1     | Disabled   | <input type="checkbox"/>            |
| 2. | Wien-1067<br>Other Provider      | #4     | Disabled   | <input type="checkbox"/>            |
| 3. | HOOMX-135<br>Other Provider      | #2     | Registered | <input checked="" type="checkbox"/> |
| 4. | HOOME.105-3064<br>Other Provider | #4     | Disabled   | <input type="checkbox"/>            |
| 5. | HOOME.156-165<br>Other Provider  | #5     | Disabled   | <input type="checkbox"/>            |
| 6. | Siggate<br>Siggate               | #6     | Disabled   | <input type="checkbox"/>            |

**Gigaset.net**

| Name        | Suffix | Status   | Active                   |
|-------------|--------|----------|--------------------------|
| Gigaset.net | #9     | Disabled | <input type="checkbox"/> |

**Fixed Line Connection**

| Name       | Suffix |
|------------|--------|
| Fixed Line | #0     |

- Telephony->Connections (configure provider)

Connection Name: Name shown in Gigaset WBM (no relationship to OpenScape Office data)

Personal Provider Data: enter the client related data here

3. IP Connection

You can replace the default name with a name or the actual phone number to distinguish this connection.

Connection Name or Number:

**Auto Configuration**

If your provider has issued you with a configuration code, you can use it to start the automatic connection configuration here.

Auto Configuration Code:

Provider:

**Personal Provider Data**

Authentication Name:

Authentication password:

Username:

Display name:

|                         |                                                                                                                    |
|-------------------------|--------------------------------------------------------------------------------------------------------------------|
| <b>Phone Value</b>      | <b>configured in OpenScape Office: Station-&gt;IP Clients-&gt;SIP Clients-&gt;<br/>Edit workpoint client data:</b> |
| Authentication Name     | Client-SIP User ID                                                                                                 |
| Authentication password | Password                                                                                                           |
|                         | <b>configured in OpenScape Office: Station-&gt;IP Clients-&gt;SIP Clients-&gt;<br/>Edit station parameters</b>     |
| Username                | Call number                                                                                                        |
| Display name            | Optional, Phone name can only be seen in the network traces, OpenScape Office uses the name configured in system   |

- Telephony->Connections Parameter / Show Advances Settings  
General Provider Data: enter the OpenScape Office IP-address  
Network: active NO and NEVER

**General Provider Data**

Domain:

Proxy server address:

Proxy server port:

Registrar server:

Registrar server port:

Registration refresh time:  sec

**Network**

STUN enabled: ☐ Yes ☒ No

STUN server:

STUN port:

STUN refresh time:  sec

NAT refresh time:  sec

Outbound proxy mode: ☐ Always ☐ Auto ☒ Never

Outbound proxy:

Outbound proxy port:

- Telephony-> Number Assignment

If more than one provider is configured on the device you **MUST** select the provider which should be used for outgoing calls. This step is not necessary if only one provider is used.

The screenshot shows the 'Number Assignment' configuration page. On the left, a sidebar lists navigation options: IP Configuration, Telephony (Connections, Audio, Number Assignment, Call Forwarding, Dialling Plans, Network Mailbox, Advanced Settings), Messaging, Services, Handsets, and Miscellaneous. The 'Number Assignment' option is highlighted. The main content area is titled 'Default Connection'. It includes settings for 'Linetype for outgoing calls' (radio buttons for VoIP and Fixed Line, with VoIP selected) and 'Automatic Fallback to Fixed Line' (radio buttons for Yes and No, with No selected). Below this is the 'Handset' section for handset 3064. It contains a table with columns for 'Connections', 'for outgoing calls', and 'for incoming calls'. The table lists several connections: BocaR.131-1362, Wien-1067, HOOMX-135 (highlighted with a red box), and HOOME.105-3064. The 'for outgoing calls' column has radio buttons, and the 'for incoming calls' column has checkboxes. HOOMX-135 has its outgoing radio button selected and its incoming checkbox checked.

- Telephony-> Advances Settings -> DTMF over VoIP

On S450IP set RFC2833 to allow sending of DTMF digits with RFC2833.

On C470 IP set either automatic (default) or RFC2833

The screenshot shows the 'DTMF over VoIP connections' configuration page. The sidebar is similar to the previous screenshot, but 'Advanced Settings' is highlighted. The main content area is titled 'DTMF over VoIP connections'. It includes a 'Send settings' section with three checkboxes: Audio, RFC 2833 (highlighted with a red box), and SIP Info. Below this is a note: 'When using G.722-Codecs (wide-band connection) DTMF Signals cannot be transmitted over audio.' The 'Call Transfer' section includes 'Use the R key to initiate call transfer with the SIP Refer method.' (radio buttons for Yes and No, with Yes selected), 'Transfer Call by On-Hook' (radio buttons for Yes and No, with No selected), 'Preferred Refer To' (radio buttons for Original URL and Target's contact info, with Target's contact info selected), and 'Automatic Refer To' (radio buttons for Yes and No, with No selected). At the bottom, a note states: 'R key settings are disabled because the R key is being used for call transfer.'

## 2.3.2 Call Forwarding

The endpoint offers

- CFB When busy
- CFNR No reply
- CFU Always

Home Settings Status Log Off

IP Configuration

Telephony

- Connections
- Audio
- Number Assignment
- Call Forwarding**
- Dialling Plans
- Network Mailbox
- Advanced Settings

Messaging

Services

Handsets

Miscellaneous

Call Forwarding

| Connection     | When      | Call number |
|----------------|-----------|-------------|
| BocaR.131-1362 | Off       |             |
| Wien-1067      | Off       |             |
| HOOMX-135      | Off       | 5140        |
| HOOME.105-3064 | Off       |             |
| HOOME.156-165  | When busy |             |
|                | No reply  |             |
|                | Always    |             |
| Sipgate        | Off       |             |
| Gigaset.net    | Off       |             |

Set Cancel

### 2.3.3 Message Waiting

Activate the MWI support at Gigaset 450/470 IP Endpoint at the Administrator menu:  
Settings Telephony → Network Mailbox

Home Settings Status

IP Configuration

Telephony

- Connections
- Audio
- Number Assignment
- Call Forwarding
- Dialling Plans
- Network Mailbox**
- Advanced Settings

Messaging

Services

Handsets

Miscellaneous

Network Mailbox

| Connection     | Call number | Active                              |
|----------------|-------------|-------------------------------------|
| BocaR.131-1362 | 1300        | <input type="checkbox"/>            |
| Wien-1067      | 1001        | <input type="checkbox"/>            |
| HOOMX-135      | 2040        | <input checked="" type="checkbox"/> |
| HOOME.105-3064 |             | <input type="checkbox"/>            |
| HOOME.156-165  |             | <input type="checkbox"/>            |
| Sipgate        |             | <input type="checkbox"/>            |
| Fixed Line     |             |                                     |

Set Cancel

### 2.3.4 Distinctive Ringing

Not supported by Gigaset

### 2.3.5 Known limitations and restrictions

## 2.4 X-lite SIP



For information see the Counterpath homepage:

<http://www.counterpath.com/x-lite.html>

Used Endpoint Software:

**X-Lite** Version 3.0 build 30942

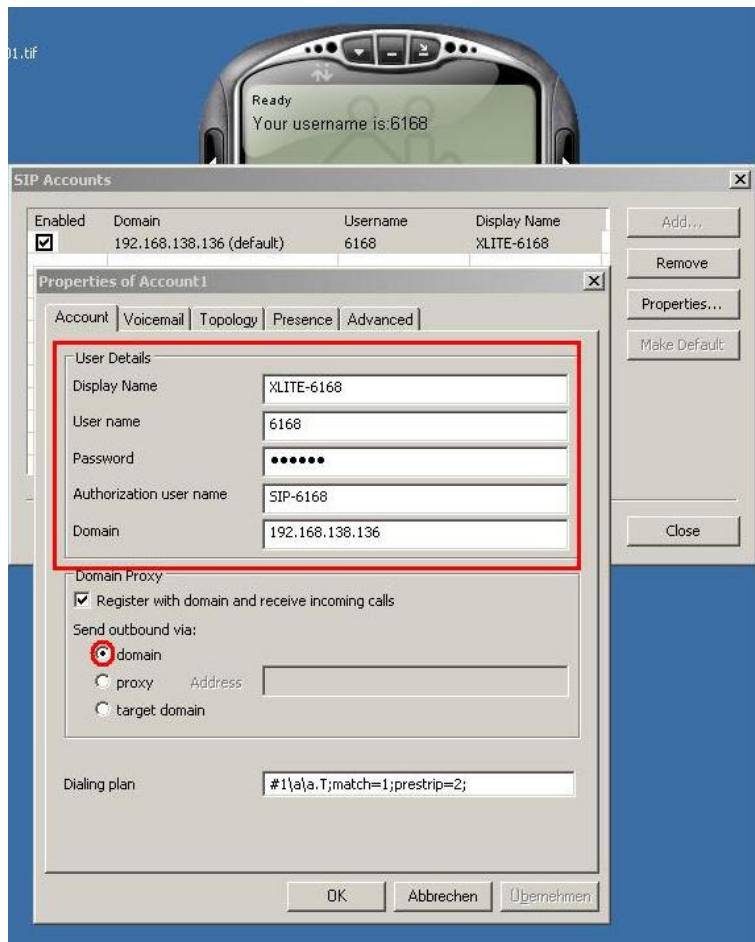
### 2.4.1 Basic Configuration

The following steps describes the configuration for the X-lite SIP Client

- Network & Registration  
Select SIP Account Setting to enter the relevant account parameter



To register the X-lite Client enter the parameter as shown in next screenshot (use the same parameter as administrated in OpenScope Office system for this Workpoint Client)



|                         |                                                                                                                  |
|-------------------------|------------------------------------------------------------------------------------------------------------------|
| Phone Value             | <b>configured in OpenScope Office: Station-&gt;IP Clients-&gt;SIP Clients-&gt; Edit station parameters</b>       |
| Display Name            | Optional, Phone name can only be seen in the network traces, OpenScope Office uses the name configured in system |
| User Name               | Call number                                                                                                      |
|                         | <b>configured in OpenScope Office: Station-&gt;IP Clients-&gt;SIP Clients-&gt; Edit workpoint client data:</b>   |
| Password                | Password                                                                                                         |
| Authorization user name | Client-SIP User ID                                                                                               |
|                         | <b>configured in OpenScope Office:</b>                                                                           |
| Domain                  | IP-Address of OpenScope Office                                                                                   |

## 2.4.2 Call Forwarding

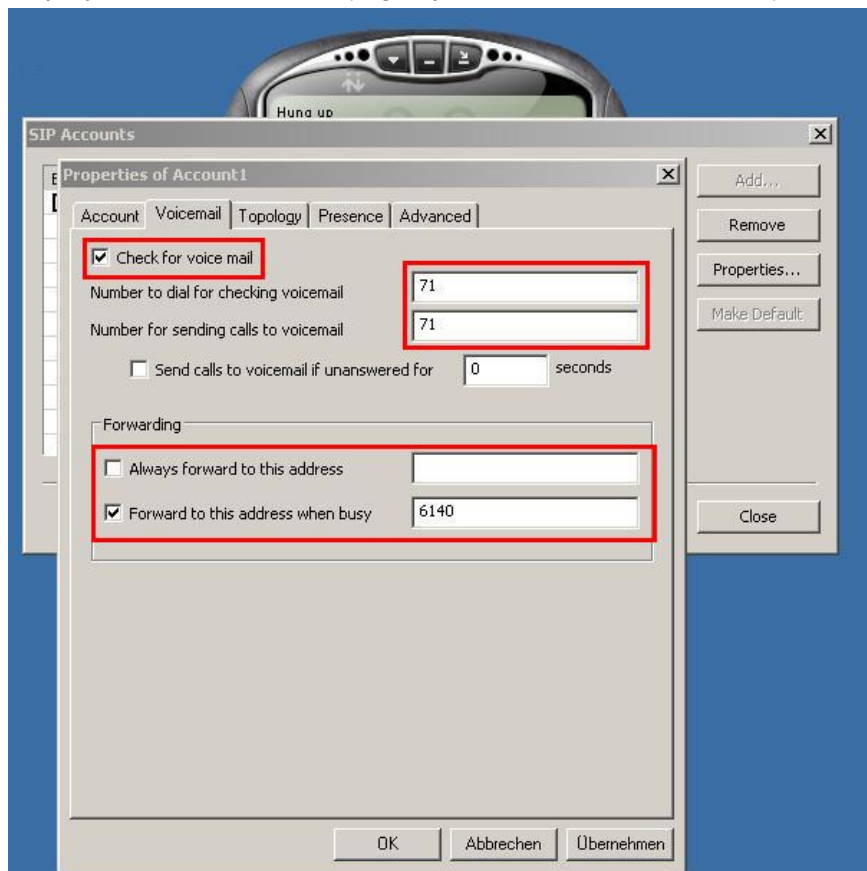
The endpoint offers

- CFB Forward to this address when busy
- CFNR Send calls to voicemail ...
- CFU Always forward to this address

Individual call forwarding targets can be configured for both services.

CFNR is supported to Voicemail only.

Forwarding targets are configured with the call number. When the entries are saved they are displayed as the SIP-URI (e.g. sip:6140@192.168.138.136)



## 2.4.3 Message Waiting

To activate the MWI service with subscription enable the voicemail support and set the voicemail call number as shown in 2.4.2.

A waiting message is signaled with a special icon  on the display of the X-lite client.

## 2.4.4 Distinctive Ringing

Not supported by X-lite Client

## 2.4.5 Known limitations and restrictions

## 2.5 Grandstream GXP280



For information see the Grandstream homepage:

[http://www.grandstream.com/products/gxp\\_series/gxp280/gxp280.html](http://www.grandstream.com/products/gxp_series/gxp280/gxp280.html)

Used Endpoint:

Produkt-Modell: GXP280 (HW0.3B)

Software Version: Programm-- 1.2.3.5 Bootloader-- 1.1.6.8

### 2.5.1 Basic Configuration

Default Administrator password: "admin"

#### Basic Settings

If no DHCP is used, enter the IP network configuration parameters as used in your network:

| Grandstream Device Configuration                                                                                                                    |                |                   |         |  |
|-----------------------------------------------------------------------------------------------------------------------------------------------------|----------------|-------------------|---------|--|
| STATUS                                                                                                                                              | BASIC SETTINGS | ADVANCED SETTINGS | ACCOUNT |  |
| End User Password: <input type="text"/> (purposely not displayed for security protection)                                                           |                |                   |         |  |
| IP Address: <input type="radio"/> dynamically assigned via DHCP (default) or PPPoE<br>(will attempt PPPoE if DHCP fails and following is non-blank) |                |                   |         |  |
| PPPoE account ID: <input type="text"/>                                                                                                              |                |                   |         |  |
| PPPoE password: <input type="text"/>                                                                                                                |                |                   |         |  |
| Host name<br>(Option 12): <input type="text"/>                                                                                                      |                |                   |         |  |
| Domain name<br>(Option 15): <input type="text"/>                                                                                                    |                |                   |         |  |
| Vendor Class ID<br>(Option 60): <input type="text" value="Grandstream GXP280"/>                                                                     |                |                   |         |  |
| Preferred DNS server: <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/>   |                |                   |         |  |
| <input checked="" type="radio"/> statically configured as:                                                                                          |                |                   |         |  |
| IP Address: <input type="text" value="192"/> <input type="text" value="168"/> <input type="text" value="138"/> <input type="text" value="193"/>     |                |                   |         |  |
| Subnet Mask: <input type="text" value="255"/> <input type="text" value="255"/> <input type="text" value="255"/> <input type="text" value="0"/>      |                |                   |         |  |
| Gateway: <input type="text" value="192"/> <input type="text" value="168"/> <input type="text" value="138"/> <input type="text" value="249"/>        |                |                   |         |  |
| DNS Server 1: <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/>           |                |                   |         |  |
| DNS Server 2: <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/>           |                |                   |         |  |

To get the correct time display set

- Daylight Saving Time
- Time Display Format
- Date Display Format
- Display Clock instead of Date

according to your needs:

|                                                                                                                     |                                                                               |
|---------------------------------------------------------------------------------------------------------------------|-------------------------------------------------------------------------------|
| Time Zone:                                                                                                          | GMT+1:00 (Paris, Amsterdam, Berlin, Rome, Vienna, Madrid, Warsaw, Brussels) ▼ |
| Allow DHCP Option 2 to override Time Zone setting:<br><input checked="" type="radio"/> No <input type="radio"/> Yes |                                                                               |
| Daylight Savings Time:                                                                                              | <input type="radio"/> No <input checked="" type="radio"/> Yes                 |
| Optional Rule: 3.2.7.2.0:11.1.7.2.0:60                                                                              |                                                                               |
| Time Display Format:                                                                                                | <input type="radio"/> 12 HOUR <input checked="" type="radio"/> 24 HOUR        |
| <input type="radio"/> Year-Month-Day                                                                                |                                                                               |
| Date Display Format:                                                                                                | <input type="radio"/> Month-Day-Year                                          |
| <input checked="" type="radio"/> Day-Month-Year                                                                     |                                                                               |
| Display Clock instead of Date:                                                                                      | <input type="radio"/> No <input checked="" type="radio"/> Yes                 |

### Advanced settings:

Enter the IP-Address of your OpenScope Office as NTP server here:

|                                                                                                               |                                    |
|---------------------------------------------------------------------------------------------------------------|------------------------------------|
| NTP Server:                                                                                                   | 192.168.138.72 (URI or IP address) |
| Allow DHCP Option 42 to override NTP server:<br><input checked="" type="radio"/> No <input type="radio"/> Yes |                                    |

### Advanced settings:

The following settings should be left in default

| Grandstream Device Configuration |                                                                                                    |
|----------------------------------|----------------------------------------------------------------------------------------------------|
| STATUS                           | BASIC SETTINGS                                                                                     |
| Admin Password:                  | (purposely not displayed for security protection)                                                  |
| G723 rate:                       | <input checked="" type="radio"/> 6.3kbps encoding rate <input type="radio"/> 5.3kbps encoding rate |
| iLBC frame size:                 | <input checked="" type="radio"/> 20ms <input type="radio"/> 30ms                                   |
| iLBC payload type:               | 97 (between 96 and 127, default is 97)                                                             |
| Silence Suppression:             | <input checked="" type="radio"/> No <input type="radio"/> Yes                                      |
| Voice Frames per TX:             | 2 (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)                                 |
| Layer 3 QoS:                     | 48 (Diff-Serv or Precedence value)                                                                 |
| Layer 2 QoS:                     | 802.1Q/VLAN Tag 0 802.1p priority value 0 (0-7)                                                    |
| Data VLAN Tag:                   | 1: 0 2: 0 3: 0 (can't use the same non-zero value as 802.1Q tag)                                   |
| No Key Entry Timeout:            | 4 (in seconds, default is 4 seconds)                                                               |
| Use # as Dial Key:               | <input type="radio"/> No <input checked="" type="radio"/> Yes                                      |
| local RTP port:                  | 5004 (1024-65400, default 5004, must be even)                                                      |
| Use random port:                 | <input checked="" type="radio"/> No <input type="radio"/> Yes                                      |
| keep-alive interval:             | 20 (in seconds, default 20 seconds)                                                                |
| Use NAT IP:                      | (if specified, this will be used in SIP/SDP message)                                               |
| STUN server:                     | (URI or IP:port)                                                                                   |

If you have to update the phone SW, provide the address of your TFTP server here. In case you want to have automatic updates enabled e.g. during reboot, set the flags accordingly.

|                                           |                                                                                                                                                                                                     |                                                                  |
|-------------------------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|------------------------------------------------------------------|
| <b>Firmware Upgrade and Provisioning:</b> | Upgrade Via                                                                                                                                                                                         | <input checked="" type="radio"/> TFTP <input type="radio"/> HTTP |
|                                           | Firmware Server Path:                                                                                                                                                                               | <input type="text" value="192.168.138.12"/>                      |
|                                           | Config Server Path:                                                                                                                                                                                 | <input type="text"/>                                             |
|                                           | Firmware File Prefix:                                                                                                                                                                               | <input type="text"/>                                             |
|                                           | Firmware File Postfix:                                                                                                                                                                              | <input type="text"/>                                             |
|                                           | Config File Prefix:                                                                                                                                                                                 | <input type="text"/>                                             |
|                                           | Config File Postfix:                                                                                                                                                                                | <input type="text"/>                                             |
|                                           | Allow DHCP Option43 and Option 66 to override server:                                                                                                                                               |                                                                  |
|                                           | <input checked="" type="radio"/> No <input type="radio"/> Yes                                                                                                                                       |                                                                  |
|                                           | Automatic Upgrade:                                                                                                                                                                                  |                                                                  |
|                                           | <input checked="" type="radio"/> No <input type="radio"/> Yes, check for upgrade every <input type="text" value="10080"/> minutes (default 7 days)                                                  |                                                                  |
|                                           | <input type="radio"/> Always Check for New Firmware<br><input type="radio"/> Check New Firmware only when F/W pre/suffix changes<br><input checked="" type="radio"/> Always Skip the Firmware Check |                                                                  |
|                                           | Authenticate Conf File:                                                                                                                                                                             |                                                                  |
|                                           | <input checked="" type="radio"/> No <input type="radio"/> Yes (cfg file would be authenticated before acceptance if set to Yes)                                                                     |                                                                  |
| <b>Phonebook XML Download:</b>            | Enable Phonebook XML Download:                                                                                                                                                                      |                                                                  |
|                                           | <input checked="" type="radio"/> No <input type="radio"/> YES, HTTP <input type="radio"/> YES, TFTP                                                                                                 |                                                                  |
|                                           | Phonebook XML Server Path:                                                                                                                                                                          | <input type="text"/>                                             |
|                                           | Phonebook Download Interval:                                                                                                                                                                        | <input type="text" value="0"/> (0-720, in minutes)               |
|                                           | Remove Manually-edited entries on Download:                                                                                                                                                         |                                                                  |
|                                           | <input checked="" type="radio"/> No <input type="radio"/> Yes                                                                                                                                       |                                                                  |
| <b>LDAP Directory:</b>                    | LDAP Script Server Path: <input type="text"/>                                                                                                                                                       |                                                                  |
| <b>Offhook Auto Dial:</b>                 | <input type="text"/> (User ID/extension to dial automatically when offhook, max length 35)                                                                                                          |                                                                  |
| <b>DTMF Payload Type:</b>                 | <input type="text" value="101"/>                                                                                                                                                                    |                                                                  |
| <b>Onhook Threshold:</b>                  | <input type="text" value="800 ms"/>                                                                                                                                                                 |                                                                  |

The following entries can be left in default (North American tones). If local tones are required this has to be changed.

|                                                                                                                            |                                                       |                                                       |
|----------------------------------------------------------------------------------------------------------------------------|-------------------------------------------------------|-------------------------------------------------------|
| <b>Distinctive Ring Tone:</b>                                                                                              | Custom ring tone 1, used if incoming caller ID is     | <input type="text"/>                                  |
|                                                                                                                            | Custom ring tone 2, used if incoming caller ID is     | <input type="text"/>                                  |
|                                                                                                                            | Custom ring tone 3, used if incoming caller ID is     | <input type="text"/>                                  |
| <b>System Ring Tone:</b>                                                                                                   | <input type="text" value="f1=440,f2=480,c=200/400;"/> |                                                       |
|                                                                                                                            | Dial Tone                                             | <input type="text" value="f1=350,f2=440;"/>           |
|                                                                                                                            | Message Waiting                                       | <input type="text" value="f1=350,f2=440,c=10/10;"/>   |
|                                                                                                                            | Ring Back Tone                                        | <input type="text" value="f1=440,f2=480,c=200/400;"/> |
| <b>Call Progress Tones:</b>                                                                                                | Call-Waiting Tone                                     | <input type="text" value="f1=440,f2=440,c=25/525;"/>  |
|                                                                                                                            | Busy Tone                                             | <input type="text" value="f1=480,f2=620,c=50/50;"/>   |
|                                                                                                                            | Reorder Tone                                          | <input type="text" value="f1=480,f2=620,c=25/25;"/>   |
| Syntax: f1=val, f2=val [, c=on1/off1[-on2/off2[-on3/off3]]];<br>(Frequencies are in Hz and cadence on and off are in 10ms) |                                                       |                                                       |

Disable not supported features, this will hide this features on the UI

|                                 |                                                               |
|---------------------------------|---------------------------------------------------------------|
| <b>Disable Direct IP Calls:</b> | <input type="radio"/> No <input checked="" type="radio"/> Yes |
|---------------------------------|---------------------------------------------------------------|

If you want to use a different language, you have to select “secondary Language” and provide the corresponding language file via TFTP. See downloadchapter

|                                                                                                                   |                                                                                                                                                                             |
|-------------------------------------------------------------------------------------------------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Display Language:                                                                                                 | <input type="radio"/> English <input type="radio"/> Chinese<br><input checked="" type="radio"/> Secondary Language <input type="text" value="ger"/> (Language File postfix) |
| <input type="button" value="Update"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/> |                                                                                                                                                                             |
| All Rights Reserved Grandstream Networks Inc. 2004-2009                                                           |                                                                                                                                                                             |

## Registration and Basic Telephony

### Account settings:

| Phone Value            | configured in OpenScope Office:                                                                                  |
|------------------------|------------------------------------------------------------------------------------------------------------------|
| SIP-Server:            | IP-Address of OpenScope Office                                                                                   |
|                        | <b>configured in OpenScope Office: Station-&gt;IP Clients-&gt;SIP Clients-&gt; Edit station parameters</b>       |
| SIP User ID:           | Call number                                                                                                      |
|                        | <b>configured in OpenScope Office: Station-&gt;IP Clients-&gt;SIP Clients-&gt; Edit workpoint client data:</b>   |
| Authenticate Password: | Password                                                                                                         |
| Authenticate ID :      | Client-SIP User ID                                                                                               |
| Name                   | Optional, Phone name can only be seen in the network traces, OpenScope Office uses the name configured in system |

| Grandstream Device Configuration                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                          |                |
|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|----------------|
| STATUS                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                    | BASIC SETTINGS |
| Account Name: <input type="text" value=""/> (e.g., MyCompany)<br>SIP Server: <input type="text" value="192.168.138.29"/> (e.g., sip.myccompany.com, or IP address)<br>Outbound Proxy: <input type="text" value=""/> (e.g., proxy.myprovider.com, or IP address)<br>SIP User ID: <input type="text" value="3760"/> (the user part of an SIP address)<br>Authenticate ID: <input type="text" value="SIP-3760"/> (can be same or different from SIP UserID)<br>Authenticate Password: <input type="password" value="*****"/> (not displayed for security protection)<br>Name: <input type="text" value="GXP280-3760"/> (optional, e.g., John Doe)<br>Use DNS SRV: <input checked="" type="radio"/> No <input type="radio"/> Yes<br>User ID is phone number: <input type="radio"/> No <input checked="" type="radio"/> Yes<br>SIP Registration: <input type="radio"/> No <input checked="" type="radio"/> Yes |                |

Send DTMF:                      disable in-audio, enable via RTP (RFC2833)

|            |                                                                                                                               |
|------------|-------------------------------------------------------------------------------------------------------------------------------|
| Send DTMF: | <input type="checkbox"/> in-audio <input checked="" type="checkbox"/> via RTP (RFC2833) <input type="checkbox"/> via SIP INFO |
|------------|-------------------------------------------------------------------------------------------------------------------------------|

Adjust the codec settings if needed:

|                                         |           |                                                |           |                                       |
|-----------------------------------------|-----------|------------------------------------------------|-----------|---------------------------------------|
| Preferred Vocoder:<br>(in listed order) | choice 1: | <input type="text" value="PCMU"/>              | choice 5: | <input type="text" value="G.726-32"/> |
|                                         | choice 2: | <input type="text" value="PCMA"/>              | choice 6: | <input type="text" value="iLBC"/>     |
|                                         | choice 3: | <input type="text" value="G.729A/B"/>          | choice 7: | <input type="text" value="G.723.1"/>  |
|                                         | choice 4: | <input type="text" value="G.722 (wide band)"/> | choice 8: | <input type="text" value="GSM"/>      |
|                                         |           |                                                |           |                                       |

## Special deployment

Change Language:

The GXP280 comes with two different languages (English,Chinese)

If you want to have a different language it has to be downloaded via TFTP.

A language pack (GXP\_Language\_Pack.zip) is available at the Grandstream download site.

<http://www.grandstream.com/firmware.html#note8>

This language pack has the compiled file which is read to be used for GXP series. Each zip file has only one particular language in it.

How to use:

1. Open the zip file
2. Open the desired language zip file
3. Copy the gxp.lpf to the TFTP server path and rename it with a postfix e.g. gxp\_ger.lpf
4. Check that your TFTP Server is running.
5. Access the advance setting of the Web UI, select Secondary Language and enter postfix e.g. "ger" without the "\_"
6. Save and reboot the phone

### 2.5.2 Hold/Retrieve/Alternate

Pressing the "Flash" key will put a call on HOLD or retrieved it from HOLD. A consultation call can be established when a call is held. Toggle/alternate can be invoked by pressing the flash key during consultation.

|   |                                                                                             |
|---|---------------------------------------------------------------------------------------------|
| ! | HOLD and all features based on HOLD will be disabled when "Send Flash Event" is set to Yes. |
|---|---------------------------------------------------------------------------------------------|

|                   |                                     |                           |
|-------------------|-------------------------------------|---------------------------|
| Send Flash Event: | <input checked="" type="radio"/> No | <input type="radio"/> Yes |
|-------------------|-------------------------------------|---------------------------|

### 2.5.3 Transfer

Attended -, Semi-Attended- and Blind Transfer is supported.

Semi Attended Transfer Mode MUST be set to "Send REFER with early dialog". If set to RFC5589 (default) the transferor will remain busy until the transfer target accepts the call.

|                              |                               |                                                               |
|------------------------------|-------------------------------|---------------------------------------------------------------|
| Semi-attended Transfer Mode: | <input type="radio"/> RFC5589 | <input checked="" type="radio"/> Send REFER with early dialog |
|------------------------------|-------------------------------|---------------------------------------------------------------|

Transfer can be disabled:.

|                   |                                     |                           |
|-------------------|-------------------------------------|---------------------------|
| Disable Transfer: | <input checked="" type="radio"/> No | <input type="radio"/> Yes |
|-------------------|-------------------------------------|---------------------------|

## 2.5.4 CLIP/CLIR/CNIP - Name and Number presentation

The phone can display names (default) or the call number

|                                     |                                                  |                                                                     |
|-------------------------------------|--------------------------------------------------|---------------------------------------------------------------------|
| Display CID instead of Name:        | <input checked="" type="radio"/> No              | <input type="radio"/> Yes                                           |
| Enable CLIR if required, by setting |                                                  |                                                                     |
| Send Anonymous                      | Yes                                              |                                                                     |
| Anonymous Method                    | Use Privacy Header                               |                                                                     |
| Send Anonymous:                     | <input checked="" type="radio"/> No              | <input type="radio"/> Yes (caller ID will be blocked if set to Yes) |
| Anonymous Method:                   | <input checked="" type="radio"/> Use From Header | <input type="radio"/> Use Privacy Header                            |

## 2.5.5 Call Waiting / Call offer

Call waiting is enabled by default in GXP280 but has to be enabled in OpenScape Office WBM. As this is a station oriented parameter there is no need to configure it in the phone. Nevertheless two parameters are provided:

|                            |                                     |                           |
|----------------------------|-------------------------------------|---------------------------|
| Disable Call-Waiting:      | <input checked="" type="radio"/> No | <input type="radio"/> Yes |
| Disable Call-Waiting Tone: | <input checked="" type="radio"/> No | <input type="radio"/> Yes |

## 2.5.6 Call Forwarding

The endpoint offers

- CFU Always Call Forwarding unconditional

CF has to be activated/deactivated on the phone via a predefined soft key

## 2.5.7 Message Waiting

For this feature the “Account Settings”

- Subscribe for MWI
- Voice Mail UserID: Access number of VM

have to be configured.

|                                   |                                     |                                      |
|-----------------------------------|-------------------------------------|--------------------------------------|
| SUBSCRIBE for MWI:                | <input type="radio"/> No            | <input checked="" type="radio"/> Yes |
| SUBSCRIBE for Registration Event: | <input checked="" type="radio"/> No | <input type="radio"/> Yes            |
| Proxy-Require:                    | <input type="text"/>                |                                      |
| Voice Mail UserID:                | <input type="text" value="71"/>     | (UserID for voice mail system)       |

A waiting message is signaled by a red light on top of the phone.

## 2.5.8 Distinctive Ringing

Not supported by GXP280

### 2.5.9 Local phone features

- DND – Do Not Disturb

The MUTE key can be used to invoke DND.

The feature can be deactivated by administration

|                     |                                     |                                                                                        |
|---------------------|-------------------------------------|----------------------------------------------------------------------------------------|
| Disable DND Button: | <input checked="" type="radio"/> No | <input type="radio"/> Yes (MUTE/DEL button pressing will have no effect if set to Yes) |
|---------------------|-------------------------------------|----------------------------------------------------------------------------------------|

- Conference

GXP280 offers a local 3 party conference. Active and held call can be connected to a 3 way conference by pressing the CONF key.

The feature can be deactivated by administration

|                     |                                     |                           |
|---------------------|-------------------------------------|---------------------------|
| Disable Conference: | <input checked="" type="radio"/> No | <input type="radio"/> Yes |
|---------------------|-------------------------------------|---------------------------|

### 2.5.10 Known limitations and restrictions

## 2.6 Grandstream GXV3140



For information see the Grandstream homepage:

[http://www.grandstream.com/products/gxv\\_series\\_phone/gxv3140/gxv3140.html](http://www.grandstream.com/products/gxv_series_phone/gxv3140/gxv3140.html)

Used Endpoint:



# GXV3140 Multimedia Phone Administration Interface

**Status** Account 1 Account 2 Account 3 Advanced Settings Maintenance Application Settings

**Status**  
Account Status  
Network Status  
**System Info**

## System Info

|                   |                        |
|-------------------|------------------------|
| Product Model     | GXV3140                |
| Hardware Revision | V0.4A                  |
| PN Code           | 9630001104A            |
| Boot Version      | 1.0.3.2                |
| Core Version      | 1.0.3.4                |
| DSP Version       | 1.0.3.25               |
| Base Version      | 1.0.3.16               |
| Program Version   | 1.0.3.24               |
| GUI-A Version     | 1.0.3.3                |
| GUI-B Version     | 1.0.3.3                |
| System Up Time    | 25 minutes, 48 seconds |

Product highlights:

3 line multimedia phone with integrated video, multimedia player, Internet radio, IM client ...

## 2.6.1 Basic Configuration

Default Administrator login “admin”, password: “admin”

The phone supports up to 3 lines to make establish calls.



To allow features like consultation or conference at least two accounts have to be configured in the phone with identical configuration parameters.  
EXCEPTION: Only for account 1 the flag SIP registration=yes is activated.

The screenshot displays the 'GXV3140 Multimedia Phone Administration Interface' with the 'Account 1' tab selected. The 'General Settings' section is active, showing various configuration fields for the account. The fields include 'Account Active' (checked), 'Account Name' (GXV3140-3364), 'SIP Server' (192.168.138.70), 'SIP User ID' (3364), 'Authenticate ID' (SIP-3364), 'Authenticate Password' (masked with dots), 'Voice Mail UserID' (71), 'Name' (GXV3140-3364), and 'User ID is phone number' (unchecked). 'Save' and 'Cancel' buttons are at the bottom.

| General Settings          |                                         |
|---------------------------|-----------------------------------------|
| Account Active :          | <input checked="" type="checkbox"/> Yes |
| Account Name :            | GXV3140-3364                            |
| SIP Server :              | 192.168.138.70                          |
| SIP User ID :             | 3364                                    |
| Authenticate ID :         | SIP-3364                                |
| Authenticate Password :   | •••••                                   |
| Voice Mail UserID :       | 71                                      |
| Name :                    | GXV3140-3364                            |
| User ID is phone number : | <input type="checkbox"/> Yes            |

For endpoints connected to the LAN NAT Traversal MUST be set to NO

The screenshot displays the 'GXV3140 Multimedia Phone Administration Interface' with the 'Account 1' tab selected. The 'Network Settings' section is active, showing fields for 'Outbound Proxy', 'DNS Mode' (A Record), 'NAT Traversal' (highlighted with a red box and set to NO), and 'Proxy-Require'. 'Save' and 'Cancel' buttons are at the bottom.

| Network Settings |          |
|------------------|----------|
| Outbound Proxy : |          |
| DNS Mode :       | A Record |
| NAT Traversal :  | NO       |
| Proxy-Require :  |          |

Configure the Account SIP settings, SIP registration and SUBSCRIBE for MWI MUST be set only for Account 1 (primary Account)

The screenshot shows the 'GXV3140 Multimedia Phone Administration Interface' with a top navigation bar including 'Status', 'Account 1' (selected), 'Account 2', 'Account 3', 'Advanced Settings', 'Maintenance', and 'Application Settings'. A left sidebar lists 'Account 1' settings: 'General Settings', 'Network Settings', 'SIP Settings' (highlighted), 'Codec Settings', and 'Call Settings'. The main area is titled 'SIP Settings' and contains the following configuration options:

- SIP Registration :** ☒ Yes **Set for 1st Account only !**
- Unregister On Reboot :** ☐ Yes
- Register Expiration (m) :** 15
- Wait Time Retry Registration (s) :** 20
- Local SIP Port :** 5060
- SUBSCRIBE for MWI :** ☒ Yes (highlighted with a red box)
- Session Expiration (s) :** 180

This section continues the 'SIP Settings' configuration for Account 1. The settings include:

- Min-SE (s) :** 90
- UAC Specify Refresher :** Omit
- UAS Specify Refresher :** UAC
- Force INVITE :** ☐ Yes
- Caller Request Timer :** ☐ Yes
- Callee Request Timer :** ☐ Yes
- Force Timer :** ☐ Yes
- Enable 100rel :** ☐ Yes
- SIP Transport :** UDP
- Symmetric RTP :** ☒ Yes
- Support SIP Instance ID :** ☐ Yes (highlighted with a red box)
- Validate Incoming Messages :** ☐ Yes
- SIP T1 Timeout :** 0.5 sec
- SIP T2 Interval :** 4 sec
- Remove OBP from route :** ☐ Yes

At the bottom of the form are 'Save' and 'Cancel' buttons.

The screenshot shows the 'Codec Settings' page for 'Account 1'. The interface includes a sidebar with navigation links: General Settings, Network Settings, SIP Settings, **Codec Settings**, and Call Settings. The main content area is titled 'Codec Settings' and contains the following fields:

- Enable Video :** A dropdown menu set to 'Yes'.
- DTMF :** Radio buttons for 'In audio', 'RFC2833' (selected), and 'SIP INFO'.
- DTMF Payload Type :** A text input field containing '101'.
- Preferred Vocoder :** Two lists of vocoders. The 'Available' list contains G723.1, G726-32, GSM, and L16-256. The 'Selected' list contains PCMU, PCMA, G729A/B, and G722.
- Preferred Video Codec :** Two lists of video codecs. The 'Available' list contains H263 and H263+. The 'Selected' list contains H264.

The dial plan has to be configured as {x+ | \*x+} to allow dialling of all strings (default dial plan).

The Refer To Use Target Contact MUST be activated to allow transfer

The screenshot shows the 'Call Settings' page for 'Account 1'. The interface includes a sidebar with navigation links: General Settings, Network Settings, SIP Settings, Codec Settings, and **Call Settings**. The main content area is titled 'Call Settings' and contains the following fields:

- Dial Plan Prefix :** A text input field.
- DialPlan :** A text input field containing '{x+ | \*x+}', highlighted with a red box.
- Early Dial :** A checkbox labeled 'Yes'.
- Refer-To Use Target Contact :** A checkbox labeled 'Yes', checked, and highlighted with a red box.
- Auto Answer :** A dropdown menu set to 'No'.
- Send Anonymous :** A checkbox labeled 'Yes'.
- Anonymous Call Rejection :** A checkbox labeled 'Yes'.

## 2.6.2 Hold/Retrieve/Alternate

Hold / retrieve is controlled by a dedicated Key : 

## 2.6.3 Transfer

Blind - and Attended-Transfer is supported

In Account->Call Settings-> Refer To Use Target Contact MUST be activated to allow Blind transfer

Blind transfer is invoked by pressing  and entering the transfer target.

For invoking Attended-Transfer please refer to the description in the user manual.

Excerpt from manual:

**Attended Transfer:** Press the "LINE" button () to select an idle line to use for attended transfer; this will place the other party on hold immediately. Dial the number that you wish to transfer to and after confirmation from the party, press the "CALL TRANSFER" button. The phone will display the following message: "Dial Number (Blind) OR Select Line (Attended)". (See figure below). Press the "LINE" button and select the line on hold.

## 2.6.4 CLIP/CLIR/CNIP - Name and Number presentation

The phone can display names (default) or the call number

Privacy can be activated by feature code and/or Web-interface

| Feature Code | Feature                                    |
|--------------|--------------------------------------------|
| *30          | Block Caller ID (for all subsequent calls) |
| *31          | Send Caller ID (for all subsequent calls)  |

## 2.6.5 Call Waiting / Call offer

Call waiting is enabled by default in GXV3140 but has to be enabled in OpenScape Office WBM too. As this is a station oriented parameter there is no need to configure it in the phone. Nevertheless two parameters are provided in Web Interface to disable call waiting:



Control of Call Waiting is possible by feature codes as well.

## 2.6.6 Call Forwarding

The endpoint offers

- CFU      **Unconditional Call Forward**
- CFB      **Busy Call Forward**
- CFNR    **Delayed Call Forward**

Call forwarding is activated/deactivated by feature codes.

| Feature Code | Feature                                                                                                                                                                |
|--------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| *72          | <b>Unconditional Call Forward:</b><br>Dial *72 + Phone/Ext. Number followed by the # key. Wait for a dial-tone and then hang up (dial-tone means input is successful). |
| *73          | <b>Cancel Unconditional Call Forward:</b><br>Dial *73 and wait for a dial-tone before hanging up.                                                                      |
| *90          | <b>Busy Call Forward:</b><br>Dial *90 + Phone/Ext. Number followed by the # key. Wait for a dial- tone and then hang up.                                               |
| *91          | <b>Cancel Busy Call Forward:</b><br>dial *91 and wait for a dial-tone before hanging up.                                                                               |
| *92          | <b>Delayed Call Forward:</b><br>Dial *92 + Phone/Ext. Number followed by the # key. Wait for a dial-tone and then hang up.                                             |
| *93          | <b>Cancel Delayed Call Forward:</b><br>Dial *93 and wait for a dial-tone before hanging up.                                                                            |

In addition a configuration via Web-Interface is possible. The timer for CFNR is configurable using the Web-interface only.

## 2.6.7 Message Waiting

For this feature the “Account Settings”

- Voice Mail UserID: Access number of VM
- Subscribe for MWI

have to be configured.

A waiting message is signaled by the blue LED on top of the phone.

Voicemail access is possible by dedicated key  if the Voice Mail UserID is configured correctly

## 2.6.8 Distinctive Ringing

Not supported by GXV3140.

The device can configure distinctive ringtones for 3 different caller IDs

## 2.6.9 Local phone features

GXV3140 offers a local 3 party conference. Active and held call can be connected to a 3 way conference by pressing the  key.

## 2.6.10 Known limitations and restrictions

Even if the phone comes with a lot of multimedia features and Web application support, the current software has some deficiencies in terms of call and feature handling.

As the phone supports up to 3 lines, features like consultation and conference are implemented by using different lines. It is not possible to invoke such features with only one line. Thus the user interface for handling such features is rather complex and needs a lot of key presses.

The phone has no easy option to configure the local tones for a specific country.

The phone needs a REBOOT for a lot of configuration changes. As it is not clear which change needs a reboot and which not it is recommended to REBOOT the phone after every configuration.



## 2.7 Nokia E52/E75/N97



E52:



**NOKIA**  
Connecting People *E75*



E75



N97:

For information see the NOKIA homepage:

<http://www.nokia.de/produkte/mobiltelefone>

Produkt-Modell: Nokia E52, E75, N97

Software Version: show firmware version on phone with “\*#0000#”

Nokia E75-(RM-412): S60, VoIP Rel 3.1, Firmware 202.12.01

Nokia E52 (RM-469): S60, VoIP Rel 3.1, Firmware 33.002(.237.03)

Nokia N97 (RM-505): S60, VoIP Rel 3.1, Firmware 21.0.045(.238.03)

SIP VoIP Settings: E52/E75: SIP\_VoIP\_3\_x\_Settings\_v2\_0\_en.sis

N97: SIP\_VoIP\_3\_1\_Settings\_S60\_5\_x\_v1\_0\_en.sis

### 2.7.1 Basic Configuration

#### Nokia E52/E75/N97 SIP client configuration:

1. Install SIP VoIP 3.x Settings application (SIP\_VoIP\_3\_x\_Settings\_v2\_0\_en.sis) on your mobile device using Nokia PC Suite.
2. *Menu-> Ctrl. Panel-> Net settings-> Advanced VoIP settings-> Create new service:*
3. Select *Create new SIP profile* option:
4. Configure *Username* "<SIP call number>@<OpenScape Office/HG1500 IP address>"
5. Configure *Password* "<password>" if authentication is configured in OpenScape Office/HiPath: UserID = <SIP call number> and Realm = <OpenScape Office/HG1500 IP address>
6. Answer following question "*Would you like to create presence settings for the service?*" with "No"
7. Select following option for "*Activate service*"
8. Now the WLAN configuration is started, if not yet done:
9. Select your WLAN network (SSID is should be displayed) and enter Preshared key (PSK)

Nokia phone does not allow editing of SIP profile settings as long as VoIP Service is active. If editing is necessary, then switch phone temporarily to Offline mode and do not allow WLAN access in Offline mode: *Menu-> Ctrl. Panel-> Profiles-> Offline* (or: *push red on hook button* and select *Offline*).

The VoIP service is activated again after switching to profile *General* and next *internet call* attempt.

It is recommended to **disable "Comfort Noise" (CN)** in SIP profile, to get better voice quality. Nokia phone shows this setting as "CN codec", that is relevant for G.711 and iLBC codec if it exists in VoIP settings.

1. Switch phone to Offline mode
2. *Menu-> Ctrl. Panel-> Net settings-> Advanced VoIP settings->VoIP services-> Select your Service-> Codecs:*
3. Select the "*CN*" codec and delete it
4. switch phone back to profile *General*.

### 2.7.2 Hold/Retrieve/Alternate

To be completed

### 2.7.3 Transfer

Attended Transfer is supported.

### 2.7.4 CLIP/CLIR/CNIP - Name and Number presentation

The phone can display names (default) or the call number

To be completed

### 2.7.5 Call Waiting / Call offer

To be completed

### 2.7.6 Call Forwarding

Not supported.

### 2.7.7 Message Waiting

Configuration for Voicemail Notification:

Nokia devices can notify the user about new voice messages in HiPath Voicemail system (OpenScape Office or IVM but not EVM). There is always a new message in inbox, when number of new messages is changing.

Voicemail must be configured, to get the notification and callback to voicemail option:

1. Switch phone to Offline mode
2. *Menu-> Ctrl. Panel-> Net settings-> Advanced VoIP settings->VoIP services-> Select your Service-> Profile settings:*
3. Select your SIP profile in *Voicemailbox Settings ID*
4. Configure *Voicemailbox address "<Voicemail call number>@<OpenScape Office/HG1500 IP address>"*
5. switch phone back to profile *General*.

### 2.7.8 Distinctive Ringing

Not supported.

### 2.7.9 Local phone features

To be completed

### 2.7.10 Known limitations and restrictions

- Nokia phone configuration MUST be done via additional VoIP setting tool
- "Comfort Noise" feature should be disabled on Nokia E52 and E75 phone; otherwise the voice may appear shortly interrupted; disruptive clicking and noise will be heard on some ITSP calls (e.g. toplink)
- Voice in direction to Nokia Phone is distinctly delayed. Reducing Jitter buffer in Nokia phone (default = 200 ms) seems not to take effect.
- some call transfer scenarios may fail
- Nokia phones does not Re-Register, when LAN connectivity to SIP Registrar is lost for more than 2 minutes but WLAN connection is active. VoIP service is then disabled on phone. There is no problem, when WLAN connectivity is lost as in standard use case "leaving WLAN home zone"!
- Nokia devices cannot be used in an environment where the Signaling and Payload encryption feature is used
- Nokia devices have payload problems if Codec G723 is used, thus this codec has to be disabled in all devices (gateways, phones) in the network

MR's related to Nokia devices

|        |                                                                      | E52 | E75 | N97 |
|--------|----------------------------------------------------------------------|-----|-----|-----|
| H74052 | call from HFA put on hold before release                             | X   | X   | X   |
| H74070 | call released when SIP is put on hold by HFA                         | X   | X   | X   |
| H74074 | TDM cannot be put on hold a second time                              | X   | X   | X   |
| H74114 | sporadically no transfer with Nokia SIP pos                          | X   | X   | X   |
| No MR  | is not doing semi attended transfer (no REFER, CANCEL after on hook) | -   | X   | -   |
| no MR  | call released when SIP is put on hold by TDM twice                   | X   | -   | x   |
| no MR  | no SIP Re-Register when IP connection lost for some minutes          | X   | -   | x   |
| no MR  | E52 sometimes not responding after answering SIP call                | X   | -   | -   |
| H77595 | No payload after blind transfer with SPE on                          |     | X   |     |
| H78641 | 1-way payload after blind transfer (G723)                            |     | X   |     |

## 2.8 Mediatrix 4102S

For information see the Mediatrix homepage:

<http://www.media5corp.com/voip-access-points/mediatrix-4100-series>

Used Endpoint:

Produkt-Modell: Mediatrix 4102S

Software Version: Dgw 2.0.7.110 (Profile: 4102-MX-D2000-44)

### 2.8.1 Basic Configuration

Default factory values for the web access are:

- **User Name:** admin
- **Password:** administrator

### Network Settings

If no DHCP is used, enter the IP network configuration parameters as used in your network:

Enter Default Gateway, DNS and SNTP under **Host** settings

Enter device IP-Address under **Interfaces**

The screenshot displays the Mediatrix web interface. At the top, there is a navigation bar with tabs: System, Network (selected), POTS, SIP, Telephony, Call Router, Management, and Reboot. Below this, a sub-menu bar includes: Status, Host, Interfaces, VLAN, QoS, Local Firewall, IP Routing, Network Firewall, NAT, and DHCP Server. The main content area is titled 'Status' and contains three sections:

- Interfaces Status:** A table with columns: Interface, Link, IP Address, Default Router, and Status. It shows one entry: Rescue (wan) with IP 192.168.138.196/24, Default Router 192.168.138.249, and Status Active.
- Host Status:** A form with sections for General Configuration, Host Name Configuration, Default Gateway Configuration, DNS Configuration, and SNTP Configuration. The values shown are: Automatic Configuration Interface: Uplink; Host Name: (none); Domain Name: (none); Default Gateway: 192.168.138.249; Primary DNS: 192.168.138.249; SNTP Host: 192.168.138.72:123.
- Firewall:** A table with columns: Scope, #, Source Address, Source Port, Destination Address, Destination Port, Protocol, Connection State, and Action. It shows 'No rule currently applied in the Firewall'.
- Ethernet Ports Status:** A table with columns: Name, Link, and Speed. It shows LAN (lan) as Disconnected and WAN (wan) as 100 Mbit/s Half-duplex.

## SIP

### SIP-Gateways

Select the correct interface port (e.g. rescue if device is connected via WAN port) and enter the port number



☐ System
 ☐ Network
 ☐ POTS
 ☒ SIP
 ☐ Telephony
 ☐ Call Router

#### ✚ Gateways

| Gateway Status |                   |      |             |       |
|----------------|-------------------|------|-------------|-------|
| Name           | Network Interface | Port | Secure Port | State |
| default        | Rescue            | 5060 | 0           | Ready |

| Gateway Configuration |                   |      |             |   |
|-----------------------|-------------------|------|-------------|---|
| Name                  | Network Interface | Port | Secure Port |   |
| default               | Rescue            | 5060 | 0           | - |
|                       |                   |      |             | + |

## SIP-Servers

Enter the OpenScope Office IP address for Registrar and Proxy



☐ System
 ☐ Network
 ☐ POTS
 ☒ SIP
 ☐ Telephony
 ☐ Call Router

#### ✚ Servers

| SIP Default Servers  |                     |  |
|----------------------|---------------------|--|
| Registrar Host:      | 192.168.138.70:5060 |  |
| Proxy Host:          | 192.168.138.70:5060 |  |
| Outbound Proxy Host: |                     |  |

| SIP Gateway Specific Registrar Servers |                  |                |
|----------------------------------------|------------------|----------------|
| Gateway Name                           | Gateway Specific | Registrar Host |
| default                                | No               | 192.168.0.10:0 |

| SIP Gateway Specific Proxy Servers |                  |                |                     |
|------------------------------------|------------------|----------------|---------------------|
| Gateway Name                       | Gateway Specific | Proxy Host     | Outbound Proxy Host |
| default                            | No               | 192.168.0.10:0 | 0.0.0.0:0           |

## Account settings:

| Phone Value                             | configured in OpenScope Office:                                                                                  |
|-----------------------------------------|------------------------------------------------------------------------------------------------------------------|
| Registrar Host                          | IP-Address of OpenScope Office                                                                                   |
| Proxy Host:                             | IP-Address of OpenScope Office                                                                                   |
| <b>configured in SIP-Registrations</b>  | <b>configured in OpenScope Office: Station-&gt;IP Clients-&gt;SIP Clients-&gt; Edit station parameters</b>       |
| User name                               | Call number                                                                                                      |
| Friendly Name                           | Optional, Phone name can only be seen in the network traces, OpenScope Office uses the name configured in system |
| <b>configured in SIP-Authentication</b> | <b>configured in OpenScope Office: Station-&gt;IP Clients-&gt;SIP Clients-&gt; Edit workpoint client data:</b>   |
| Password                                | Password                                                                                                         |
| User Name                               | Client-SIP User ID                                                                                               |

Enter UserName and Friendly name, select enable for Register and assign the Gateway

Registrations

| Endpoint   | User Name | Gateway Name | Registrar           | Status     |
|------------|-----------|--------------|---------------------|------------|
| Phone-Fax1 | 3364      | default      | 192.168.138.70:5060 | Registered |
| Phone-Fax2 | 3365      | default      | 192.168.138.70:5060 | Registered |

| User Name | Gateway Name | Registrar | Status |
|-----------|--------------|-----------|--------|
|-----------|--------------|-----------|--------|

| Endpoint   | User Name                         | Friendly Name                        | Register                              | Gateway Name                         |
|------------|-----------------------------------|--------------------------------------|---------------------------------------|--------------------------------------|
| Phone-Fax1 | <input type="text" value="3364"/> | <input type="text" value="MX-4102"/> | <input type="button" value="Enable"/> | <input type="text" value="default"/> |
| Phone-Fax2 | <input type="text" value="3365"/> | <input type="text" value="MX-4102"/> | <input type="button" value="Enable"/> | <input type="text" value="default"/> |

| Index | User Name | Gateway Name                     |
|-------|-----------|----------------------------------|
|       |           | <input type="button" value="+"/> |

Submit Submit & Refresh Registration

Click Edit to get to the web page for entering the authentication data for the endpoint. Validate Realm SHOULD be left "disabled"

Authentication

| Priority | Apply To | Endpoint   | Gateway | Validate Realm | Realm | User Name | Actions                                                                                                 |
|----------|----------|------------|---------|----------------|-------|-----------|---------------------------------------------------------------------------------------------------------|
| 1        | Endpoint | Phone-Fax1 |         | Disable        |       | SIP-3364  | Edit <input type="button" value="v"/> <input type="button" value="+"/> <input type="button" value="-"/> |
| 2        | Endpoint | Phone-Fax2 |         | Disable        |       | SIP-3365  | Edit <input type="button" value="v"/> <input type="button" value="+"/> <input type="button" value="-"/> |
| 3        | Unit     |            |         | Enable         |       |           | Edit <input type="button" value="v"/> <input type="button" value="+"/> <input type="button" value="-"/> |
| 4        | Unit     |            |         | Enable         |       |           | Edit <input type="button" value="v"/> <input type="button" value="+"/> <input type="button" value="-"/> |
| 5        | Unit     |            |         | Enable         |       |           | Edit <input type="button" value="v"/> <input type="button" value="+"/> <input type="button" value="-"/> |
| 6        | Unit     |            |         | Enable         |       |           | Edit <input type="button" value="v"/> <input type="button" value="+"/> <input type="button" value="-"/> |
| 7        | Unit     |            |         | Enable         |       |           | Edit <input type="button" value="v"/> <input type="button" value="+"/> <input type="button" value="-"/> |
| 8        | Unit     |            |         | Enable         |       |           | Edit <input type="button" value="v"/> <input type="button" value="+"/> <input type="button" value="-"/> |

Authentication

| Priority | Apply To                              | Endpoint                                | Gateway                       | Validate Realm                       | Realm                         | User Name                             | Password                           |
|----------|---------------------------------------|-----------------------------------------|-------------------------------|--------------------------------------|-------------------------------|---------------------------------------|------------------------------------|
| 1        | <input type="text" value="Endpoint"/> | <input type="text" value="Phone-Fax1"/> | <input type="text" value=""/> | <input type="text" value="Disable"/> | <input type="text" value=""/> | <input type="text" value="SIP-3364"/> | <input type="text" value="*****"/> |

Cancel Submit Submit & Refresh Registration

## Telephony - CODECS

To enable DTMF transmission the DTMF Transport MUST be set to Out-of-Band using RTP

## CODECS

Select Endpoint: Default

| CODEC         | Voice   | Data    | Advanced |
|---------------|---------|---------|----------|
| G.711 a-Law   | Enable  | Enable  | Edit     |
| G.711 u-Law   | Enable  | Enable  | Edit     |
| G.726 16Kbps  | Disable |         | Edit     |
| G.726 24Kbps  | Disable |         | Edit     |
| G.726 32Kbps  | Disable | Disable | Edit     |
| G.726 40Kbps  | Disable | Disable | Edit     |
| G.729         | Enable  |         | Edit     |
| T.38          |         | Enable  | Edit     |
| Clear Mode    | Disable | Disable | Edit     |
| Clear Channel | Disable | Disable | Edit     |
| X CCD         | Disable | Disable | Edit     |

|                                               |                       |
|-----------------------------------------------|-----------------------|
| <b>Misc</b>                                   |                       |
| <b>Jitter Buffer</b>                          |                       |
| Level:                                        | Normal                |
| Minimum:                                      | 30                    |
| Maximum:                                      | 125                   |
| <b>DTMF Transport</b>                         |                       |
| Transport Method:                             | Out-of-Band using RTP |
| Payload Type:                                 | 96                    |
| <b>Generic Voice Activity Detection (VAD)</b> |                       |
| Enable (G.711 and G.726):                     | Conservative          |

## Telephony - Misc

This selection is important for local tones

## Misc

|                    |          |
|--------------------|----------|
| <b>Country</b>     |          |
| Country Selection: | Germany1 |

Submit

## 2.8.2 Hold/Retrieve/Alternate

Hold/Retrieve is supported by the device and controlled with "Hook-Flash". No configuration is necessary to active the feature. The feature can be deactivated through the Telephony-Services menu page.

Alternate is NOT possible

## 2.8.3 Transfer

Attended- and Blind Transfer are supported by the device, no configuration is necessary. The features can be deactivated through the Telephony-Services menu page.

## 2.8.4 CLIP/CLIR/CNIP - Name and Number presentation

This depends on the connected phone; the adapter itself provides configurable CLIP.

## 2.8.5 Call Waiting / Call offer

Supported by the device, no configuration is necessary. The feature can be deactivated through the Telephony-Services menu page.

## 2.8.6 Call Forwarding

CFU, CFB and CFNR are supported by the device.

For each kind of CF an activation/deactivation string has to be defined. When defining these strings please take care of the predefined “DTMF Maps”.

**Mediatrix®**

System Network POTS SIP Telephony Call Router

DTMF Maps CODECS Call Forward Services Tone Customization Music

**Call Forward**

Select Endpoint: Default

| Call Forward On Busy          |        | Default |
|-------------------------------|--------|---------|
| Allow Activation Via Handset: | Enable |         |
| DTMF Map Activation:          | *13    |         |
| DTMF Map Deactivation:        | **13   |         |

| Call Forward On No Answer     |        |  |
|-------------------------------|--------|--|
| Allow Activation Via Handset: | Enable |  |
| DTMF Map Activation:          | *12    |  |
| DTMF Map Deactivation:        | **12   |  |
| Timeout:                      | 5000   |  |

| Call Forward Unconditional    |        | Unit Defaults |
|-------------------------------|--------|---------------|
| Allow Activation Via Handset: | Enable |               |
| DTMF Map Activation:          | *11    |               |
| DTMF Map Deactivation:        | **11   |               |

## 2.8.7 Message Waiting

Not supported.

## 2.8.8 Distinctive Ringing

Not supported

## 2.8.9 Local phone features

- Conference

Mediatrix 4102 offers a local 3 party conference. Active and held call can be connected to a 3 way conference by pressing the Flash-key. (Hook-Flash)

The feature can be deactivated by administration

## 2.8.10 Known limitations and restrictions

## 2.9 Aastra 6739i



For information see the Aastra homepage:

<http://www.aastra.de/cps/rde/xchg/SID-3D8CCB6A-2DEDB780/aastra-detewe/hs.xsl/38707.htm>

Used Endpoint:

**Aastra**

6739i

Log Off

**Status**  
System Information

**Operation**  
User Password  
Softkeys and XML  
Directory  
Reset

**Basic Settings**  
Preferences  
Account Configuration

**Advanced Settings**  
Network  
Global SIP  
Line 1  
Line 2  
Line 3  
Line 4  
Line 5  
Line 6  
Line 7  
Line 8  
Line 9  
Action URI  
Configuration Server  
Firmware Update  
TLS Support  
802.1x Support  
Troubleshooting

**System Information**

**Network Status**

| Attribute   | LAN Port | PC Port |
|-------------|----------|---------|
| Link State  | Up       | Down    |
| Negotiation | Auto     | Auto    |
| Speed       | 100Mbps  | n/a     |
| Duplex      | Full     | Half    |

**Hardware Information**

| Attribute       | Value             |
|-----------------|-------------------|
| MAC Address:    | 00-08-5D-13-DC-17 |
| BT MAC Address: | 00-08-5D-13-DC-18 |
| Platform        | 6739i Revision 0  |

**Firmware Information**

| Attribute             | Value                |
|-----------------------|----------------------|
| Firmware Version      | 3.0.1.2015           |
| Firmware Release Code | SIP                  |
| Boot Version          | 3.0.0.167            |
| Date/Time             | May 28 2010 05:18:54 |

**SIP Status**

| Line | SIP Account              | Status     | Backup Registrar Used? |
|------|--------------------------|------------|------------------------|
| 1    | 3365@192.168.138.70:5060 | Registered | No                     |
| 2    | 3365@192.168.138.70:5060 | Registered | No                     |
| 3    | 3365@192.168.138.70:5060 | Registered | No                     |

Product highlights:

3 line phone with touchscreen and programmable soft keys

### 2.9.1 Basic Configuration

Default Administrator login “admin”, password: “22222”

#### Basic Settings

In Basic Settings->Preferences some global configuration parameters must be set.

As the systems does no differ between the different lines of the phone the “Key Mode” shall be set to “Phone”. Intercom is not supported and thus shall be deactivated.

**Aastra** 6739i  
Log Off

**Status**  
System Information

**Operation**  
User Password  
Softkeys and XML  
Directory  
Reset

**Basic Settings**  
Preferences  
Account Configuration

**Advanced Settings**  
Network  
Global SIP  
Line 1  
Line 2  
Line 3  
Line 4  
Line 5  
Line 6  
Line 7  
Line 8  
Line 9  
Action URI  
Configuration Server  
Firmware Update  
TLS Support  
802.1x Support  
Troubleshooting

**Preferences**

**General**

Local Dial Plan: x+#+x+\*

Send Dial Plan Terminator: ☐ Enabled

Digit Timeout (seconds): 4

Call Waiting: ☒ Enabled

Play Call Waiting Tone: ☒ Enabled

Stuttered Dial Tone: ☒ Enabled

Message Waiting Indicator Line: All

DND Key Mode: Phone

Call Forward Key Mode: Phone

**Outgoing Intercom Settings**

Type: Off

Prefix Code:

Line: 1

**Incoming Intercom Settings**

Auto-Answer: ☐ Enabled

Microphone Mute: ☐ Enabled

Play Warning Tone: ☐ Enabled

Allow Barge In: ☐ Enabled

Ring Tones must be set according to your contry:

**Troubleshooting**

**Ring Tones**

Tone Set: Germany

Global Ring Tone

Line 1: Global

Line 2: Global

Line 3: Global

Line 4: Global

Line 5: Global

Line 6: Global

Line 7: Global

Line 8: Global

Line 9: Global

**Priority Alerting Settings**

Enable Priority Alerting: ☒ Enabled

Group: Normal ringing

External: Normal ringing

Internal: Normal ringing

Emergency: Normal ringing

Priority: Normal ringing

Auto call distribution: Normal ringing

Community 1: Normal ringing

For a correct time and date display enter the OpenScape Office IP-Address for Time Server and set TimeFormat to your needs.

Select the appropriate Input language for the phone

The screenshot shows the configuration interface for a phone. It includes several sections:

- Directed Call Pickup Settings:** Includes 'Directed Call Pickup' (checkbox, disabled) and 'Directed Call Pickup by Prefix' (text input).
- Auto Call Distribution Settings:** Includes 'Auto Available' (checkbox, disabled) and 'Auto Available Timer' (text input, value 60).
- Time and Date Setting:** This section is highlighted with a red box. It includes:
  - 'Time Format' (dropdown, value 24h)
  - 'Date Format' (dropdown, value DD MMM YYYY)
  - 'NTP Time Servers' (checkbox, checked, value Enabled)
  - 'Time Server 1' (text input, value 192.168.138.70)
  - 'Time Server 2' (text input, value 2.aastra.pool.ntp.org)
  - 'Time Server 3' (text input, value 3.aastra.pool.ntp.org)
- Language Settings:** This section is also highlighted with a red box. It includes:
  - 'Webpage Language' (dropdown, value English)
  - 'Input Language' (dropdown, value Deutsch)
  - 'Language 1', 'Language 2', 'Language 3', and 'Language 4' (text inputs, all empty).

A 'Save Settings' button is located at the bottom left of the configuration area.

## Advances Settings

On this page the registration data for the phone is entered

| Phone Value           |                                                                                                                  |
|-----------------------|------------------------------------------------------------------------------------------------------------------|
| Screen Name           | Displayed on the phone, no protocol relevance                                                                    |
| Screen Name 2         | Displayed on the phone, no protocol relevance                                                                    |
|                       | <b>configured in OpenScape Office: Station-&gt;IP Clients-&gt;SIP Clients-&gt; Edit station parameters</b>       |
| Phone number:         | Call number                                                                                                      |
|                       | <b>configured in OpenScape Office: Station-&gt;IP Clients-&gt;SIP Clients-&gt; Edit workpoint client data:</b>   |
| Password:             | Password                                                                                                         |
| Authentication name : | Client-SIP User ID                                                                                               |
| Caller-ID             | Optional, Phone name can only be seen in the network traces, OpenScape Office uses the name configured in system |
|                       | <b>configured in OpenScape Office:</b>                                                                           |
| Proxy-Server:         | IP-Address of OpenScape Office                                                                                   |
| Proxy-Port:           | SIP-Port, default 5060                                                                                           |

**67391**  
 Log Off

**Status**

System Information

**Operation**

User Password

Softkeys and XML

Directory

Reset

**Basic Settings**

Preferences

Account Configuration

**Advanced Settings**

Network

Global SIP

Line 1

Line 2

Line 3

Line 4

Line 5

Line 6

Line 7

Line 8

Line 9

Action URI

Configuration Server

Firmware Update

TLS Support

802.1x Support

Troubleshooting

### Global SIP Settings

**Basic SIP Authentication Settings**

|                     |             |
|---------------------|-------------|
| Screen Name         | Aastra-3365 |
| Screen Name 2       | SD33 - LAB  |
| Phone Number        | 3365        |
| Caller ID           | Aastra-3365 |
| Authentication Name | SIP-3365    |
| Password            | *****       |

|            |         |
|------------|---------|
| BLA Number |         |
| Line Mode  | Generic |

**Basic SIP Network Settings**

|                       |                |
|-----------------------|----------------|
| Proxy Server          | 192.168.138.70 |
| Proxy Port            | 5060           |
| Backup Proxy Server   | 0.0.0.0        |
| Backup Proxy Port     | 0              |
| Outbound Proxy Server | 0.0.0.0        |
| Outbound Proxy Port   | 0              |

|                         |                |
|-------------------------|----------------|
| Registrar Server        | 192.168.138.70 |
| Registrar Port          | 5060           |
| Backup Registrar Server | 0.0.0.0        |
| Backup Registrar Port   | 0              |
| Registration Period     | 0              |
| Conference Server URI   |                |

If MWI is used activate Subscription here

**Advanced SIP Settings**

Explicit MWI Subscription

Explicit MWI Subscription Period

MWI for BLA account

AS-Feature-Event Subscription

AS-Feature-Event Subscription Period

Send MAC Address in REGISTER Message

Send Line Number in REGISTER Message

Session Timer

T1 Timer

T2 Timer

Transaction Timer

Transport Protocol

Local SIP UDP/TCP Port

Local SIP TLS Port

Registration Failed Retry Timer

Registration Timeout Retry Timer

Registration Renewal Timer

BLF Subscription Period

ACD Subscription Period

BLA Subscription Period

Blacklist Duration

Park Pickup Config

Whitelist Proxy

XML SIP Notify

|                                      |                                             |
|--------------------------------------|---------------------------------------------|
| Explicit MWI Subscription            | <input checked="" type="checkbox"/> Enabled |
| Explicit MWI Subscription Period     | 86400                                       |
| MWI for BLA account                  | <input type="checkbox"/> Enabled            |
| AS-Feature-Event Subscription        | <input type="checkbox"/> Enabled            |
| AS-Feature-Event Subscription Period | 3600                                        |
| Send MAC Address in REGISTER Message | <input type="checkbox"/> Enabled            |
| Send Line Number in REGISTER Message | <input type="checkbox"/> Enabled            |
| Session Timer                        | 0                                           |
| T1 Timer                             | 0                                           |
| T2 Timer                             | 0                                           |
| Transaction Timer                    | 4000                                        |
| Transport Protocol                   | UDP                                         |
| Local SIP UDP/TCP Port               | 5060                                        |
| Local SIP TLS Port                   | 5061                                        |
| Registration Failed Retry Timer      | 1800                                        |
| Registration Timeout Retry Timer     | 120                                         |
| Registration Renewal Timer           | 15                                          |
| BLF Subscription Period              | 3600                                        |
| ACD Subscription Period              | 3600                                        |
| BLA Subscription Period              | 300                                         |
| Blacklist Duration                   | 300                                         |
| Park Pickup Config                   |                                             |
| Whitelist Proxy                      | <input type="checkbox"/> Enabled            |
| XML SIP Notify                       | <input type="checkbox"/> Enabled            |

For a good interworking with HFA phones, enter supported codecs here.  
For best interworking with other devices set Packetization Interval to 20ms.

The screenshot shows the 'RTP Settings' section of a configuration window. The 'RTP Port' is set to 3000. 'Force RFC2833 Out-of-Band DTMF' is checked and 'Enabled'. 'DTMF Method' is set to 'RTP'. 'RTP Encryption' is set to 'SRTP Disabled'. Below this is the 'Codec Preference List' section with a note: 'Note: Basic Codecs Include G.711u (8K), G.711a (8K), G.729'. There are ten codec slots. 'Codec 1' is set to 'Basic' and 'Codec 2' is set to 'G.722', both highlighted with red boxes. 'Codec 3' through 'Codec 10' are all set to 'None'. Below the codec list, the 'Packetization Interval' is set to '20', also highlighted with a red box. 'Silence Suppression' is checked and 'Enabled'. The 'Autodial Settings' section is partially visible at the bottom, showing 'Autodial Number' and 'Autodial Timeout' set to '0'. A 'Save Settings' button is at the bottom left.



If a IP-DECT system is installed in your network Packetization Interval MUST be set to 20ms

## 2.9.2 Hold/Retrieve/Alternate

Hold / retrieve is controlled by a soft key. Alternate is controlled by the line keys

## 2.9.3 Transfer

Blind -, SemiAttended- and Attended-Transfer is supported and controlled via soft keys

## 2.9.4 CLIP/CLIR/CNIP - Name and Number presentation

The phone can display names (default) or the call number  
Privacy can not be activated

## 2.9.5 Call Waiting / Call offer

Call waiting is enabled by default in Aastra6739i but has to be enabled in OpenScape Office WBM too. As this is a station oriented parameter there is no need to configure it in the phone. Nevertheless two parameters are provided in Basic Settings->Preferences to disable call waiting.

## 2.9.6 Call Forwarding

The endpoint offers

- CFU **Call Forward All**
- CFB **Call Forward Busy**
- CFNR **Call Forward No Answer**

It can be configured on the phone (i-key or programmed soft key).

### **2.9.7 Message Waiting**

For this feature the “Account Settings”

- Subscribe for MWI

have to be configured.

A waiting message is signaled by a message on the screen and a blinking red LED at the “message key”.

There is no configuration for the callback access of the voicemail server, this has to be programmed on a softkey

### **2.9.8 Destinctive Ringing**

Not supported by Aastra6739i.

### **2.9.9 Local phone features**

Aastra6739i offers a local 3 party conference. Active and held call can be connected to a 3 way conference by pressing the Conf soft key.

### **2.9.10 Known limitations and restrictions**

Registration problem: can be “patched with configuration parameter:

`sip contact matching: 3`

#### About Siemens Enterprise Communications Group (SEN Group)

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